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Smart Multiple Access Topologi Employing Intelligent Physical Layer for addressing the Mobile Node

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SMART MULTIPLE ACCESS TOPOLOGY EMPLOYING INTELLIGENT PHYSICAL LAYER FOR ADDRESSING THE MOBILE NODES

DISSERTATION
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ELECTRONIC SYSTEMS
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7. R. Sanyal, E. Cianca, and R. Prasad, "Beyond LTE: next generation multiple access technology with intelligent lower layers," in Proceedings of ICWN 2011, the World Congress Computer Science, Computer Engineering and Applied Computing and International Conference, Las Vegas, USA,2011,p474
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9. Rajarshi Sanyal, 'Challenges in Interoperability and Roaming between LTE - Legacy core for Mobility Management, Routing, Real Time Charging ' ITU WORLD 2011, Geneva, Switzerland, 26th October 2011, published in IEEE Explore.
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12. Rajarshi Sanyal,' Framework for Realizing mobile and computer communication through color signals' ,SAM 2006, World Congress in Computer Science , Computer Engineering and Applied Computing, Las Vegas,USA.
13. Rajarshi Sanyal, Mobile and Computer Communications Through Colour Signals - An Approach Note. WINSYS 2006: 73-78.Portugal.

This thesis has been submitted for assessment in partial fulfillment of the PhD degree. The thesis is based on the submitted or published scientific papers which are listed above. Parts of the papers are used directly or indirectly in the extended summary of the thesis. As part of the assessment, co-author statements have been made available to the assessment committee and are also available at the Faculty. The thesis is not in its present form acceptable for open publication but only in limited and closed circulation as copyright may not be ensured.

Abstract

The factors that will drive the ontogenesis of next generation mobile networks are namely the evolution of the mobile devices and ever incremental demand for data traffic triggered by the proliferation of smart phones and the dongles. Apprehending an exponential growth in numbers of the mobile users, the cell size of the mobile network will shrink. Eventually, there will be an upsurge of signaling messages interchanged between the device and the network due to an increase in handover related operations. Attempt should be made to reduce the interaction with the network and simplify the mobility and location management related processes. We also consider in this dissertation the ever increasing demand for data traffic caused by the new generation mobile devices; we set out a technical architecture of a mobile network to abridge the intricate mobility and location management processes as well as the addressing principle to render more operational efficiency and cater to the needs of the data hungry applications that we envisage for the next generation devices and networks. Our primary objective is to propose a technique of wireless communication where we render intelligence to the physical layer to take part in some processes which is otherwise confined to higher layer signalling activities, like, for example, addressing of a node. If we have a method to actuate user identification for the purpose of mobility management by implementing lower layer processes, we can simplify or eliminate the layer 7 processes and hence reduce the intricacy and costs of mobile network elements and user equipment.

In the State-of-the-Art mobile network, the handset initiates a control message in a given periodicity to update its location towards the Network. The Handover process requires exchange of substantial control channel messages over the so called ‘signalling channels’. These network procedures guzzle processing power in the handset/network, signalling resources and also drain out battery in the device.

The smart modulation scheme presented in this thesis abets the identification of the network users right at the physical layer and relieves the application layer from the prosaic activities related to addressing and mobility to a considerable extent. Noticeably, in all existing mobile network topologies, modulation schemes are designed solely for conveying data symbols for all users in the complex plane (user identification is not possible in layer 1). We propose to drift from this paradigm and actuate user identification and mobility at physical layer. As a direct consequence, the incessant signalling inter-exchange between the network and the devices reduces manifold, as the network does not need to continuously track the mobile station within the coverage area. This perks up the handset performance too.

Nano mobile devices designed for ‘Human to Human’ or ‘Machine to Machine’ applications have inviolable constraints to operate in a mobile network, due to limited power resource and restrictions in RF power output. With the State-of-the-Art mobile network, as it stands today, optimised operation of a nano mobile device is untenable, because it desecrates considerable resources solely to maintain its presence in the network. A mobile network based on smart location management and addressing scheme provides an elegant solution to address these basic needs and helps in fostering this new breed of mobile devices. The access network design is simplified with less resource hungry processes associated to mobility management. This is aligned with our endeavour to make the mobile networks cleaner, greener and leaner.

For a typical closed user group type of network, we propose a multiple access mechanism and network topology which will eliminate the involvement of intelligent core network adjuncts in the network area, and to use this intelligent physical layer to directly reach any node over the air interface in the coverage area. Some typical implementations may be automotive communication systems, corporate networks, campus networks, railway networks etc., which use closed numbering scheme and coverage within the community network. This system is ‘presence agnostic’, hence the network is

not aware of the exact location of the mobile node in the network area. Therefore, the mobility management process is not envisaged. For a basic closed user network, the design does not follow the cellular topology. The network area is represented by a single Location area.

The smart modulation scheme termed as SMNAT (Smart Mobile Network Access Topology) has been proposed for global mobile network and for Device to Device Communication envisaged for 5G. Further, we had also delved in the various application scenarios, namely Machine to Machine communications Intelligent Automotive Communications.

The objective of this research is to propose an architectural alternative to 4G / 5G on the basis of apprehending the various pain points related to the State-of-the-Art and to solve them by eventually realising a new mobile network framework, a new multiple access methodology and a smart mobility management model.

Wireless communication based on colour signals is another form of ‘physical layer network addressing’ which has been proposed in subsequent chapters. The implementation scenarios set out in this thesis are Wireless Local Loop (WLL) communications and voice transcoding.

Related to this research work, we have 7 conference papers (World Congress USA, IEEE Conferences in Europe, ITU Conference in Geneva), 2 patents (1 granted, 1 pending) and 5 published peer reviewed papers in European and American Journals. 1 more paper is expected to be published shortly.

Our research has been able to garner interest of telecom community (GSMA) and was featured by BBC Technology news during Sept 2011.

Main Contributions of the thesis

- To identify the issues of mobility management in next generation networks due to the interplay between the network and manifold devices , like M2M , Nano mobile devices, Automotive mobile devices as well as the smartphone used by humans.
- To address the issue of the signalling storm and to propose a Smart Multiple Access Scheme to alleviate the mobility management process and reduce the signalling interaction between the devices and the network
- To propose a Mobility Management Scheme following the lines of the previous objective
- The next generation mobile network paradigm would not only attain high bandwidth per user, but would also optimise the network processes by trimming down the interaction between the mobile device and the network.
- To propose a method of communication and transcoding using electronic colour signals (pertaining to the physical plane).

The scope of the study is to bring out a global architecture for mobile communications beyond 4G where we focus on realisation of a smarter access methodology and mobility management framework. The main contribution is related to the possibility to address the user at physical layer. We used various mathematical techniques to work out the spectral efficiency, capacity, physical losses and hence propose an appropriate coding scheme for signalling and data.

Dansk Resume

De faktorer, der vil drive ontogenesis af næste generation af mobile netværk er nemlig udviklingen af de mobile enheder, og nogensinde gradvist stigende efterspørgsel efter datatrafik udløst af udbredelsen af smartphones og dongles. Pågribe en eksponentiel vækst i antallet af mobile brugere, vil cellen størrelse af mobilnettet skrumpes. Til sidst vil der være en bølge af signaleringsmeddelelser byttes mellem enheden og netværket på grund af en stigning i handover relaterede aktiviteter. Forsøg bør gøres for at reducere interaktionen med netværket og forenkle mobilitet og placering management processer. Vi mener også i denne afhandling den stadigt stigende efterspørgsel efter data trafik på grund af de nye generation af mobile enheder; vi fastsætter en teknisk arkitektur et mobilnet til forkorte de udviklede mobilitet og placering management processer samt adressering princip at gøre mere operationelle effektivitet og imødekomme behovene hos de data sultne applikationer, som vi forudsiger for de næste generation af enheder og netværk. Vores primære målsætning er at foreslå en teknik til trådløs kommunikation, hvor vi gør intelligens til det fysiske lag til at deltage i nogle processer, som ellers begrænset til højere lag signalsystemer aktiviteter, som for eksempel, adressering af en knude. Hvis vi har en metode til at aktivere brugeren identifikation med henblik på mobilitet forvaltning ved at gennemføre lavere lag processer, kan vi forenkle eller fjerne lag 7 processer og dermed reducere de forviklinger og omkostninger mobile netelementer og bruger udstyr.

I stadiet mobilnet, starter håndsettet en kontrol budskab i en given periodicitet til at opdatere sin placering mod netværket. Overdragelsen kræver udveksling af væsentlige kontrol kanal beskeder via de såkaldte signalering kanaler ". Disse netværk procedurer tykke regnekraft i håndsettet / netværk, signalering ressourcer, og også løbe ud batteriet i enheden.

Den smarte graduering præsenteres i denne afhandling abets identifikation af brugerne af nettet til højre ved det fysiske lag og lindrer applikationslaget fra prosaiske aktiviteter i forbindelse med adressering og mobilitet til et betydeligt omfang. Mærkbart, i alle eksisterende mobil netværkstopologier, graduering ordninger er udelukkende beregnet til transport datasymboler for alle brugere i komplekse plan (bruger-id er ikke muligt i lag 1). Vi foreslår at drifte fra dette paradigme og aktivere bruger-id og mobilitet på fysiske lag. Som en direkte konsekvens, den uophørlige signalering inter-udveksling mellem netværket og enhederne reducerer manifold, da netværket ikke nødt til konstant at spore mobil station inden for dækningsområdet. Dette frynsegoder røret ydeevne også.

Nano mobile anordninger til 'menneske til menneske "eller" maskine til maskine' applikationer har ukrænkelige begrænser at operere i et mobilt netværk, på grund af begrænset magt ressource og begrænsninger i RF-udgangseffekt. Med State-of-the-Art mobilnet, som det står i dag, optimeret drift af en nano mobile enhed er uholdbar. Bare fordi det vanhelliger betydelige ressourcer udelukkende opretholde sin tilstedeværelse i netværket. En mobil netværk baseret på intelligent placering ledelse og adressering giver en elegant løsning til at løse disse basale behov og hjælper med at fremme denne nye race af mobile enheder. Netadgang design er forenklet med mindre ressource sultne processer i forbindelse med mobility management. Dette er på linje med vores bestræbelser på at gøre det mobile netværk renere, grønnere og slankere.

For en typisk lukket brugergruppe type netværk, foreslår vi en multiple access mekanisme og netværkstopologi der vil fjerne inddragelsen af intelligente kerne netværk supplementer i netværket området, og til at bruge denne intelligente fysiske lag for direkte at komme enhver node via air interface i dækningsområdet. Nogle typiske implementeringer kan være automotive kommunikationssystemer, firmanetværk campus netværk, jernbanenettet mv, der anvender lukkede nummersystem og dækning inden for EF-nettet. Dette system er "tilstedeværelse agnostiker« dermed netværket ikke er klar over den nøjagtige placering af det mobile knudepunkt i netværket område.

Derfor er mobiliteten management processen ikke planlagt. For en grundlæggende lukket bruger nettet, er designet ikke følge den cellulære topologi. Netværket Området er repræsenteret ved et enkelt sted område.

Er blevet foreslået smart graduering betegnes som SMNAT (Smart Mobile Network Access Topologi) til global mobil netværk og til Enhed til enhed Kommunikation planlagt for 5G. Desuden havde vi også dykket i de forskellige anvendelsesområder scenarier, nemlig maskine til maskine kommunikation Intelligent Automotive Communications. Formålet med denne forskning er at foreslå en arkitektonisk alternativ til 4G / 5G på grundlag af pågribelsen af de forskellige smerte punkter relateret til State-of-the-Art, og at løse dem ved med tiden at realisere en ny mobil netværk rammer en ny multipel adgang metode og en intelligent mobilitet forvaltningsmodel. Trådløs kommunikation baseret på farvesignaler er en anden form for "fysiske lag netværk adressering", som er blevet foreslået i de efterfølgende kapitler. Scenarierne implementering fastsat i denne afhandling er trådløse abonnentnet (WLL) kommunikation og voice omkodning.

Relateret til dette forskningsarbejde, vi har 7 konference papers (World Congress USA, IEEE konferencer i Europa, ITU i Geneve), 2 patenter (1 indrømmet, 1 behandling) og 5 publicerede peer reviewed artikler i europæiske og amerikanske tidsskrifter. 1 flere papirer forventes at blive offentliggjort om kort tid.

Vores forskning har været i stand til at samle interesse telecom samfund (GSMA) og blev præsenteret af BBC nyheder om teknologi i løbet af september 2011.

Vigtigste Bidrag af specialet

- At identificere spørgsmål af mobility management i næste generations net på grund af samspillet mellem netværket og mangfoldige enheder som M2M, Nano mobile enheder, Automotive mobile enheder samt smartphone brugt af mennesker.
- For at løse spørgsmålet om signalering storm og foreslå en smart Multiple Access Scheme at afhjælpe mobilitet management processen og reducere signalering samspillet mellem enheder og netværk
- At foreslå en Mobility Management Ordningen efter linjerne i den tidligere målsætning
- Den næste generation af mobile netværk paradigme ville ikke blot opnå en høj båndbredde per bruger, men også optimere nettet processerne ved at trimme ned samspillet mellem den mobile enhed og netværket.
- At foreslå en metode til kommunikation og omkodning ved hjælp af elektroniske farvesignaler (vedrørende det fysiske plan). Omfanget af undersøgelsen er at bringe en global struktur for mobilkommunikation uden 4G, hvor vi fokuserer på realisering af en smartere adgang metode og mobility management rammer. Det vigtigste bidrag er relateret til muligheden for at behandle brugeren på fysiske lag. Vi brugte forskellige matematiske teknikker til at træne de spektrale effektivitet, kapacitet, fysiske tab og dermed foreslå en hensigtsmæssig kodning ordning til signalering og data.

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List of Acronyms

- ASR : Answer Supervision Rate
- AP : Access point
- AAA: Authentication , Authorisation , Accounting
- AFCRN : Absolute Radio Frequency Channel Number
- AP : Access Point
- BSSAP: Base Station System Application Part
- BTS : Base Transceiver Station
- CN : Core Network
- CCH: Common Control Channel
- CAPS : Call Attempts per second
- CSCF : Call Server Control Function (for IMS networks)
- DCH: Dedicated Control Channel
- FTE : Full Time Equivalent
- GIS: Geographical Information System
- GTP : GPRS Tunnelling protocol
- HSS : Home Subscription System
- HLR : Home Location Register
- IMS: IP Multi Media System
- IMSI : International Mobile Subscriber Identity
- ICIC : Inter Cell Interference Coordination
- IVC : Intelligent Vehicular Communication
- LTE : Long Term Evolution
- MME : Mobility Management Entity (analogue of a MSC in LTE Network)
- M2M : Machine to Machine
- M2A : Machine to Application
- MVNO : Mobile Virtual Network Operator
- MIN: Mobile Identification Number
- MSISDN: Mobile Subscriber ISDN number
- NTP : Network Time Protocol
- NID: Network ID
- QOS: Quality of Service
- RAN : Radio Access Network
- SIP : Session Initiation protocol (for VOIP)
- SID : System ID
- UE: User Equipment
- WLL: Wireless Local Loop

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1

Introduction

1.0. Background

A study by the World Bank titled "Information and Communications for Development 2012" provided a statistics that three-quarters of the world's people have access to a mobile device. The number of worldwide mobile subscriptions grew from less than 1 billion in 2000 to over 6.7 billion today (vide Chart 1). "Mobile communications offer major opportunities to advance human and economic development -- from providing basic access to health information to making cash payments, spurring job creation, and stimulating citizen involvement in democratic processes," World Bank Vice President for Sustainable Development Rachel Kyte said in a press statement.

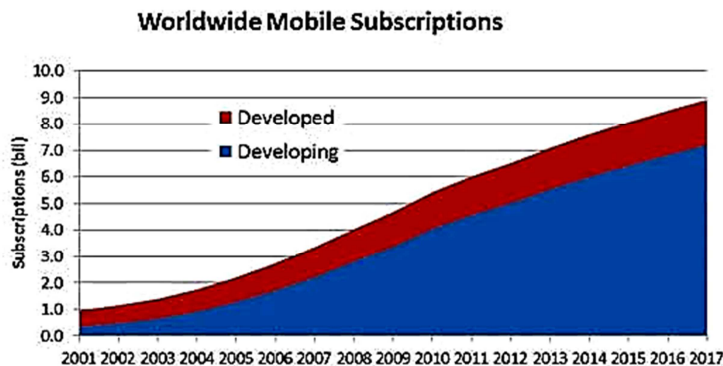


Chart 1 Global Mobile Penetration (source ITU World Telecommunications/ICT Indicators database)

Close to 5 billion of the subscriptions are in developing countries, and mobile subscriptions in low- and middle-income countries increased by more than 1,500 percent between 2000 and 2010, from 4 to 72 per 100 inhabitants. This is equal to around 67 percent of the world's population. Chart 2 & Chart 3 indicate some details of the evolution of mobile penetration across the developed and developing world and across the continents.

2 Introduction

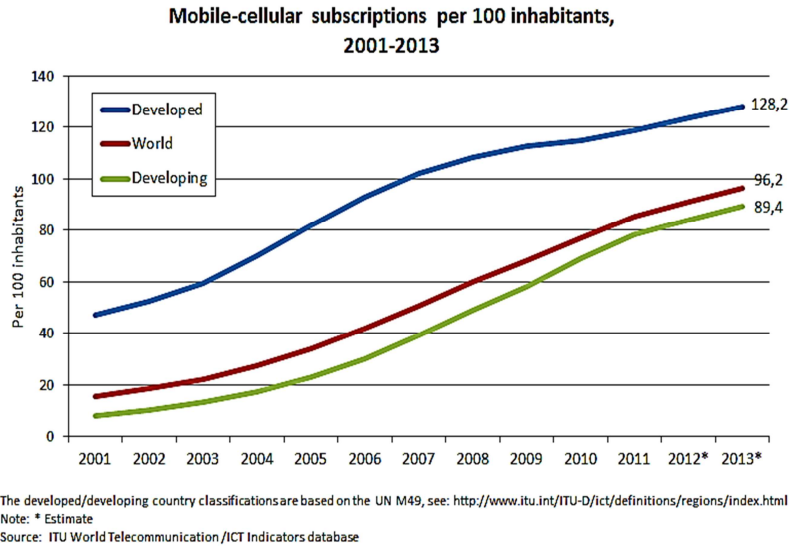


Chart 2 Distribution of Mobile penetration across the developed and developing world

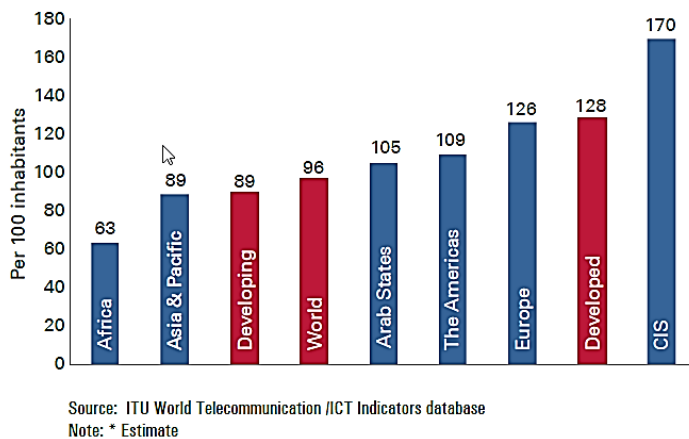


Chart 3 Distribution of Mobile penetration across the continents

The concentrations of the mobile users, especially in the developing countries, are in the urban regions.

Some of the key proponents for high urban cellular user density impact on cellular mobility, distribution and penetration are

- Industrial growth and centralisation/ agglomeration of industry
- Social Mobility,
- Enterprise and corporate mobility
- Floating work force

Catalysts for Industrial growth are:

- ▶ Economic liberalization policies can increase the degree of spatial centralization of resources and spatial concentration of manufacturing in large metropolitan areas.
- ▶ Talent / human resources required for the industry are centralized in the metros in many cases
- ▶ Economies of agglomeration → Infrastructure, availability of raw materials, transport facilities, proximity of customer and supplier.
- ▶ Chart 4 is a dashboard highlighting the primary components for industrial growth which is one of the factors for growth of mobile subscribers.

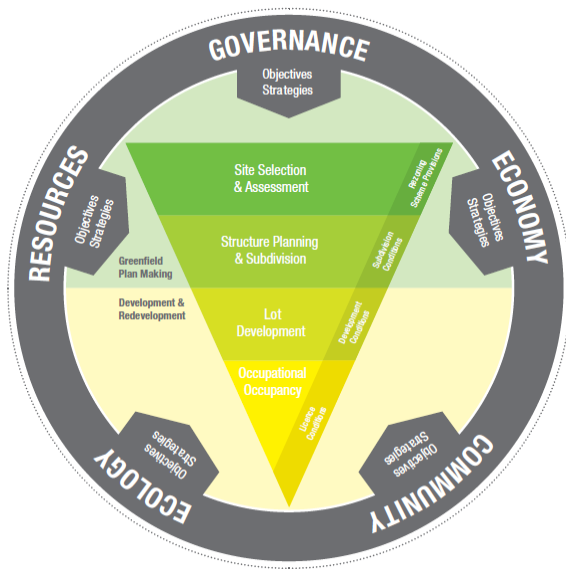


Chart 4 Proponents for industrial growth.

Social Mobility and Urbanisation:

- ▶ Social mobility measures the degree to which people's social status changes between generations.
- ▶ More children born in poorer families can, in later life, elevate their social status by achieving economic success by virtue of their own talents and motivation.
- ▶ As a result, in many cases the poorer families living in the rural regions migrate to metros for a better living, and to work in the industrial regions.
- ▶ This trend is prevalent especially in the developing countries (China, India) due to wage gap and it directly contributes to mobile user clusterisation in metros.
- ▶ Demonstration effect / lifestyle changes

Enterprise Mobility:

- ▶ Enterprise mobility is the trend toward a shift in work habits, with more employees working out of the office and using mobile devices and cloud services to perform business tasks.
- ▶ The term refers not only to mobile workers and mobile devices, but also to the mobility of corporate data. An employee may upload a corporate presentation from his or her desktop PC to a cloud storage service, then access it from a personal iPad to show at a client site, for example.

4 Introduction

- ▶ Enterprise mobility can improve employee productivity, work life balance of the employees, decrease operational cost like optimisation of office space, power , logistics etc.
- ▶ Enterprise mobility can help in reducing urban cellular clusterisation.

Floating work force:

- ▶ In the metros, there may exist a floating population.
- ▶ They are mainly factory workers / corporate employees travelling towards their work place everyday.
- ▶ As they all use the mobile devices during the day , these may lead to congestion in the cellular network.
- ▶ Meticulous network planning by the cellular companies need to be done to which takes into account this factor and to avoid a logjam in the network.

Technical challenges and adaptation in mobile networks due to the high proliferation of devices:

- ▶ Serve more users per cell → actuate better frequency reuse, deploy more cells smaller in sizes
- ▶ Serve more traffic volumes per carrier → attain higher spectral efficiency
- ▶ More cell to cell movement of user → Optimisation of cell handover procedures
- ▶ Crunch of addresses → Addition of users imply more mobile numbers. So number management needs to be optimised.
- ▶ Higher data rate requirement per user and network due to user growth, advent of new generation smartphones, social networking through mobile devices → implementation of new multiple access techniques like OFDMA (Orthogonal Frequency Division Multiple Access).

Apart from the factors mentioned above contributing to elevation of the global subscriber base, is a new breed of equipment called Machine to Machine cellular device which we find everywhere in the modern world. Some examples are the vending machines, electric meters, Point of Sales (PoS), automobiles.

Due to high penetration of mobile users, the cell sizes will shrivel. Hence the location and management and handover related messages will increase. This has a collateral impact on the bandwidth, spectral efficiency, capital and operational expenditure of the network.

All the emerging technologies aim to attain a higher data speed with the available media resources. The basic network design however remains the same. We see heavy use of signaling, complex mobility management processes and intricate honeycomb cellular architecture. With the advent of IMS and LTE there is an attempt to move towards all IP architecture [10][23][24], but the basic network processes are too close to the predecessors. Following the evolution trail from 2G to 4G, we do not discern a significant philosophical drift or focus towards simplification of the basic processes related to mobility and location management. We see the service logic for handover, frequency reuse, location updates /cancellations in 4G as compared to 2G [18][25]] have the same roots. With all the existing and the upcoming technologies, we do not perceive a significant improvisation to simplify the mobility management process and reduce the control signaling between the device and the network in order to ward of the possibility of a signaling storm which is brewing up in the LTE (4G) Networks.

Hence we propose a network architecture based on Smart Multiple Access topology which renders intelligence to the lower layers of the network .This relieves the Application layer from handling the Mobility management and addressing functions. It directly employs the physical layer to actuate the same.

All the mobile network technologies, namely GSM, UMTS, W-CDMA, CDMA and LTE are based on various blends of Multiple Access techniques. Some examples are Time division, code division, Orthogonal Frequency division, but in all those concepts the data plane carry symbols destined towards the different users in the network plane. Typically, location management and addressing are done by the higher layer, namely layer 7 of OSI, called the application layer.

With all these evolving technologies, the barriers are fixed. We have the Shannon bound with the permissible Signal-to-Noise ratio, same physical effects to distort the signal, spectrum and speed constrains. Most of the new generation technologies still attempt to conceive higher and higher data speed within the Shannon bound. The multiple access technologies that are used are fixed. With all the evolving technologies, we still have the CDMA, OFDMA (for LTE), TDMA and so on. With all these available multiple access technologies, we manipulate the way the symbols are carried over the complex plane, mainly with the attempt to enhance throughput.

We attempt to drift from this basic philosophy. We believe that if we find a way to engage the access mechanism not only to carry information for all the users, but also to render the intelligence to identify and address the users, the complexity at the higher layers is reduced to a considerable extent. The design of the network elements and the mobile nodes is simplified which also contributes to reduce the set up cost. Our research work aims to unfold a new mobile network technology based on a unique multiple access technique and a new access and core network topology.

With this technology, we thrive to shift some specific functionalities pertaining to layer five and seven mobility towards the lower layers, namely layer two and one. Hence we make the physical and data link layer more intelligent and involve them directly in the task of addressing the nodes. While we make the lower layers more intelligent, the tasks / processing by the higher layer diminish. This reduces the processing power required in the access and core layer network equipments making them cheaper. Also makes the interaction between these networks less resource intensive and overall reduces the complexity of network design and operations.

In this research proposal, we present this new technology and highlight its benefits over the existing topologies. We also investigate the problematic and cost ineffective areas of the present mobile technologies and provide an analysis how we can draw the benefits on these aspects from the proposed technology.

We will arrange the thesis according to the following chapters.

1.1. Organisation of Chapters

The thesis is arranged in the following different chapters

1. Problem statement , objectives , and motivation of the research work
2. The Smart Multiple Access Technology
3. Deploying the Novel Multiple Access Scheme for a Global Mobile Roaming Network.
4. Channel Impairments and the proposal of an appropriate error correction scheme
5. Colour Pixel Multiple Access for Wireless Local Loop deployment
6. Multisource Voice coding technology using colour coding technique for High Definition voice coding for mobile networks.
7. Conclusion and future scope

1.2. Brief Description of the chapters:

Chapter 2: Problem statement, related work, motivation and objectives

Description: All the mobile network topologies from 2G to LTE- Advanced share a common philosophy for access network design either on TDM (2G) or IP(4G) . They have the same mechanism of addressing and location management. Handover principles are comparable and the handover success rate depends on various factors including frequency allocation in the cell and the frequency reuse factor. These processes are the key to the QOS, Call Maturation rate, Efficiency and success rate for handover. It demands careful network introspection, design and operations which is a resource centric activity. The complex framework of control plane for the signalling operations consumes considerable channel bandwidth and processing power in the network devices. All this contributes to an increase in cost of the Network setup and its operations.

This chapter attempts to study the various factors that convolutes the design, implementation and operations of a present day mobile network, and paves the way for proposing a new network topology, which is fundamentally different, easy to implement and operate and also brings new promises for making the next generation services affordable to all.

Chapter 3: The Smart Mobile Network Access Topology

We discuss here the novel multiple access scheme, Smart Mobile Network Access Topology (SMNAT) and its implementation in light of different network topologies. The multiple access schemes in the ITU and ANSI domain use a common methodology of conveying the user information by the symbols in the complex plane. The proposed methodology however drifts away from this philosophy, where we also involve the symbols partly for the purpose of addressing and identifying the node in the network area. This novel multiple access methodology finds suitable application in different network topologies, starting from closed user networks, small sized networks to global roaming networks [1][15]. In this chapter, we discuss the implementation scenarios in closed user group networks with fixed addressing mechanism. The chapter also focuses on the feasibility study of implementation in scaled down mobile networks, like campus networks, railways or petroleum plants.

Chapter 4: Deploying the Novel Multiple Access Scheme (SMNAT) for a Global Mobile Roaming Network and for Device to Device Communication

In this chapter, we discuss the application of the Multiple Access scheme for a Global Roaming Network [1][15]. Mobility and the location management is the key to the success of roaming. The chapter introduces a new cellular design, frequency allocation and reuse per cell, and new handover mechanism. Here, we see the geometrical shape of the cell has changed and we propose a simple black and white checker board approach for Network design and frequency allocation. Our endeavour is to render simplicity, enhance performance and to reduce the associated network processes and signalling overhead. We discuss the architectural concepts for a global mobile network in light of this novel multiple access technologies. Following this, we study the implementation feasibility for Machine-to-Machine devices, intelligent vehicular communication network and nano mobile devices.

Further we investigate how SMNAT can be a proponent for Device to Device (D2D) communication which is one of the primary differentiators of 5G. The existing 5G research groups proposes techniques to realise D2D communication for the devices in proximity, may be in the same cell or in neighbouring cells. However SMNAT may make it possible to actuate D2D communication even

when the paired devices are separated by a distance and located in cells which may belong to different location areas.

Chapter 5: Channel Impairments and the proposal of an appropriate error correction scheme

By this time, we understand the potential and the benefits that we can derive out of this new technology compared to LTE and 2G/3G. But unlike the other technologies, it is also subject to the physical effects and will be prone to errors if the right solution for error correction is not chosen.

This chapter is meant for discussing the subject related to the channel impairment due to the physical effects, like reflection, scattering and fading. This happens for all types of mobile networks, but the impact is far-fetched with the proposed network topology than compared to the present one [1][15]. This is because the multiple access scheme directly takes part in addressing for the mobile node, and any misinterpretation of symbols in this plane should be avoided. This can jeopardise the practicality of such a multiple access scheme. Hence we propose the appropriate error correction and coding mechanisms to render robustness and accuracy and make the technology viable for implementation and operations for mobile networks.

Chapter 6: Colour Pixel Multiple Access for Wireless Local Loop deployment

Our research will remain incomplete if we do not cover the technologies which are based on the last mile access, namely the WLL. We propose a unique Multiple Access mechanism called Colour pixel Multiple Access [3][17], which can be deployed over the existing cable TV framework. It uses the same concept of deploying the symbols for addressing a user. The only difference being that the symbols of a complex plane are represented by colour dots (with a Hue and Saturation level) with fixed coordinates in a colour circle. Here each user will be represented by a colour level (Hue and Saturation level) and a specific colour pixel coordinate, reutilising the existing digital colour broadcast technology. Colour is directly used for addressing. It is also used to carry voice and data by utilising the variation of the amplitude levels (saturation levels). This unique patented technology brings vast promises in communication technology and we begin with the implementation aspect in the purview of WLL Network. This chapter discusses the details of this technology and proposes the WLL network Architecture in light of this new technology.

Chapter 7: Multisource Voice coding technology using colour coding technique for High Definition voice coding for mobile networks.

This chapter addresses a mechanism for multisource voice coding technology using colour coding technique (as proposed in chapter 5) for high definition voice transmission [4]. The audio information is mapped into colour information with the help of the intelligence derived from a phase vocoder. Each pixel in the video frame, meant for the transmission of the audio signals over video, pertains to a specific frequency in the time domain. A couple of harmonic frequencies along with the fundamental frequency are required to be processed for carrying the voice intelligence. Hence, for conveying a speech signal for a given user, a number of colour pixels are utilised. This implies that the number of users that can be accommodated depends on the resolution that is supported by the video encoding scheme and the QOS needed for carrying the voice. At the source, we actuate phase vocoding and colour pixel multiplexing. This part can be termed as channel coding as we encode each harmonic in a suitable pixel and thereby render redundancy to the original information. The video stream which comprises number of pixels per frame pertaining to the resolution is multiplexed at the source and then transmitted using an appropriate lossless source coding scheme.

Chapter 8: Conclusion and future scope

The following section lists our papers published/presented on this topic in International journals and International Conferences. The technology was featured by BBC Technology News, the reference to which has also been included.

1.3. References:

Conference Paper Details	Primary Reference in chapter	How it is used in the chapter
1. Rajarshi Sanyal, Ramjee Prasad, and Ernestina Cianca, Beyond LTE: Next Generation Multiple Access Technology with Intelligent Lower Layers. 2011 International Conference on Wireless Networks (July 18-21, 2011, Las Vegas, USA). World Congress Computer Science, Computer Engineering and Applied Computing at USA a. Ref : http://dblp.uni-trier.de	3	The description of the global roaming network in light of the Multiple Access Technology is described in this paper
2. Rajarshi Sanyal , Ernestina Cianca, Ramjee Prasad 'Rendering Intelligence at Physical Layer for Smart Addressing and Multiple Access', presented at IARIA conference , France, on wireless networks 2010 and published at IEEE digital library. http://ieeexplore.ieee.org/xpl/login.jsp?tp=&arnumber=5474027&url=http%3A%2F%2Fieeexplore.ieee.org%2Fxppls%2Fabs_all.jsp%3Farnumber%3D5474027	2	The Intelligent Physical layer and its application for a CUG network is defined in this paper
3. Rajarshi Sanyal , Ernestina Cianca, Ramjee Prasad, Colour Pixel Multiple Access- A multiple Access Technology for Next Generation Mobile Networks' at International Conference on Wireless Networks (ICWN 09),2009 World Congress Computer Science, Computer Engineering and Applied Computing at USA a. Ref : http://dblp.uni-trier.de	5	The Colour Pixel Multiple Access technology is defined in this section. In the thesis chapter, we however focus more on its application in WLL technology
4. Rajarshi Sanyal ,Transmission of Audio signals from multiple sources riding on single lossless streaming video channel using Phase vocoder by applying Joint Source Channel Coding technique, .Communication In Computers (CIC07): 2007 World Congress in Computer Science , Computer Engineering and Applied Computing in Las Vegas USA. a. Ref : http://dblp.uni-trier.de	6	The application of the colour Pixel Multiple Access technology High Definition vocoding is covered by this paper and chapter.
5. Rajarshi Sanyal, ' Framework for Realizing mobile and computer communication through color signals' 2006 World Congress in Computer Science , Computer Engineering and Applied Computing in Las Vegas , USA. a. Ref : http://dblp.uni-trier.de	5	The basic framework of Colour Pixel Multiple Access technology where the layer 2 is used for both addressing and data communication is defined.
6. Rajarshi Sanyal, Mobile and Computer Communications Through Colour Signals - An Approach Note. WINSYS 2006 : 73-78 , Portugal	5	The basic framework of Colour Pixel Multiple Access technology and its implementation in mobile network is covered here. However, the thesis chapter focuses more on its implementation in the WLL networks.
7. Rajarshi Sanyal , 'Challenges in Interoperability and Roaming between LTE - Legacy core for Mobility Management, Routing, Real Time Charging ' ITU WORLD 2011 , presented in Geneva on 26th October 2011 , published in IEEE Explore. http://ieeexplore.ieee.org/xpl/freeabs_all.jsp?arnumber=6100941	1	Referred in the chapter to highlight the interoperability issues between LTE and 2G/3G.

9 Introduction

Journal Paper Details	Primary Reference in chapter	How it is used in the chapter
1.Sanyal, R., Cianca, E. and Prasad, 'Smart mobile networks with intelligent physical layer actuating mobility management', Int. J. Mobile Network Design and Innovation, Inderscience Publishers 2012 Vol.4, No.2, pp.91 - 108	3	The description of the global roaming network in light of the Multiple Access Technology in described in this paper
2 Sanyal, R., Cianca, E. and Prasad, R 'Nano mobile network based on Smart Location Management and Addressing', Int. J. Mobile Network Design and Innovation, Inderscience, Vol. 4, No. 3, 2012	1 & 3	Issues with Nano mobile devices operating within the State-of-the-Art mobile network are discussed in chapter 1, The description of the global roaming network in light of the Multiple Access Technology in described in chapter 3
3 Sanyal, R., Cianca, E. and Prasad, R, 'Challenges for Coexistence of Machine to Machine and Human to Human Applications in Mobile Network: Concept of Smart Mobility Management, International Journal of Distributed Sensor Networks, Hindawi publishers, Volume 2012 (2012), Article ID 830371, 12 pagesdoi:10.1155/2012/830371.	1 & 3	Related work and Issues with M2M devices operating within the State-of-the-Art mobile network are discussed in chapter 1, The description of the global roaming network in light of the Multiple Access Technology in described in chapter 3
4 Sanyal, R., Cianca, E. and Prasad, 'Closed User Group Automotive communication network based on addressing at physical layer', International Journal of Interdisciplinary Telecommunications and Networking. IGI Global publishers USA, Volume 4, Issue 4. , 2012.	1 & 2& 3	Related work and Issues with IVC devices operating within the State-of-the-Art mobile network are discussed in chapter 1, IVC within CUG Network is defined in Chapter 2.The description of the global roaming network in light of the Multiple Access Technology in described in chapter 3
5. Sanyal, R., Prasad, Enabling Cellular Device to Device Data Exchange on WISDOM 5G by Actuating Cooperative Communication Based on SMNAT, International Journal of Interdisciplinary Telecommunications and Networking. IGI Global publishers USA, Volume 6, Issue 3,2014	3	Description of 5G landscape and deployment of SMNAT for WISDOM 5G
6. Sanyal, R., Cianca, E. and Prasad, 'Novel WLL architecture based on Colour Pixel Multiple Access implemented on a Terrestrial Video Network as the overlay', International Journal of Interdisciplinary Telecommunications and Networking, IGI Global publishers USA Volume 5, Issue 1,2013	5	The basic framework of Colour Pixel Multiple Access technology and its implementation in mobile network is covered here. However, the thesis chapter focuses more on its implementation in the WLL networks

Patents	Primary Reference in chapter	How it is used in the chapter
Sanyal, Rajarshi : "Framework for Realizing Mobile Network Through Color Signals" Patent App. No. 0163/MUM/2006, Govt. of India, February 2006).	5	The basic framework of Colour Pixel Multiple Access technology and its implementation in mobile network is covered here. However, the thesis chapter focuses more on its implementation in the WLL networks
Sanyal, Rajarshi : "A system of wireless networking of computers in the UHF", Patent Document (Pat. No. 188052, Govt of India, October 1995)		

News and Media

This is the future calling

By Mark Ward Technology correspondent, BBC News
<http://www.bbc.co.uk/news/technology-15079004>

2

Problem statement, related work, motivation and Objectives

2.0. Introduction and problem statement

‘Simplicity is the ultimate sophistication’.
Leonardo da Vinci

The 5th and 6th generation of mobile networks are envisioned to realize the bandwidth required for WWW (Wireless Worldwide Wide Web). They should be highly dynamic in nature and actuate self-optimization of resources to support the bandwidth for ever increasing customer base. Spectrum crunch is one of the major issues. Hence these networks should strive to achieve very high spectral efficiency and less Carrier to Interference ratio (C/I), relative to the existing technologies like LTE and WIMAX. Data speed per user should also not be compromised.

In brief, the key challenges for the 5th and 6th generation of the mobile technologies are the following.

- High data rate over large coverage areas and dense demographics.
- Reduce cost of network infrastructure and operation
- Meet the bandwidth, latency, QOS requirement for supporting the next generation network and device applications.
- Reduce power consumption in the network and the device
- Lessen the complexity in the access (radio) and core network layers.

Living through the advancements in mobile network technology from 2G and then through 3G – UMTS and finally to LTE (4G), we notice that the primary emphasis is to enable higher bandwidth per user required to drive the next generation network and device applications.

With the evolution of the mobile networks as we find from 2G to 4G [18][25], we attempt to inject more symbols (data) per unit time. This has been made possible by the evolution of the modulation schemes from BPSK in case of 2G, to 64 QAM for LTE-Advanced.

In the current Mobile Radio Access Network [9], [20], [21] there are a host of network processes, tightly synchronized and orchestrated by intelligent network elements, such as:

- Handover management
- Location management
- Call drop off management
- Interoperability and downward compatibility management
- Service control (like roaming control)
- Feature management

These processes involve complex signaling operations across the radio and the core network. The attempt to simplify the network in IMS [10][23] and LTE[24] is focused to make the core and access networks all IP. Following the evolution trail from 2G to 4G, we do not discern a significant philosophical drift towards simplification in terms of the service logic related to mobility and location management. We see a similar process for handover, frequency reuse, location updates and cancellations in 4G as compared with 2G. Hence the network elements of the State-of-the-Art still need to be equipped with the intelligence and processing power to handle all these complex signaling operations. This has a knock on effect on the handsets too. It involves considerable amount of processing power, thus leading to more battery drainage. So it is imperative to look beyond the State-of-the-Art and conceive a technology which is devoid of these constraints and is ideal for the next generation mobile devices. It is worth mentioning the Telenor's research to address the issues of mobility management in next generation networks [25].

The same goes for core network design. Mobility and location management in the core network necessitates a substantial volume of signaling interaction between the HLR/HSS (Home Location Register / Home Subscription Service) and the MSC/VLR (Mobile Switching Center/ Visited Location Register). Inefficient and un-optimised engineering and design of these Network Elements impacts the service assurance, and finally degrades to the overall QoS offered by the network. Also, a substantial amount of resources are used up for addressing the wireless node in the cellular network, both from the perspective of available bandwidth and also the associated processing activities carried out in the different network elements at the Radio Access Network (RAN) and Core Network (CN).

Managing a mobile network is hugely resource and knowledge intensive, primarily because of the inherent complexity of the network architectures. For example, during cell planning we need to consider the traffic demand to cover a specific region, availability of base station sites, available channel capacity at each base station, and the service quality at various potential traffic demand areas. The allocation of the right frequency at the cells to get an optimum frequency reuse factor is crucial to achieve smooth handovers, avoidance of call drops during handover, and overall elevate the performance of the network. These are part of the overall cell planning activity and it requires a lot of resources, in terms of FTE (Full Time Equivalent) and hence it increases the operational cost.

The switching (time/ space) matrix of a Mobile Switching Center of a 2G/3G network has a finite limit to make and break the number of calls. The Busy Hour Call Attempt handling capacity of a Switch depends much on engineering and dimensioning of this matrix. With the CSCF (Call Session Control Function) for IMS (IP Multimedia Subsystem) [10][23] and the MMEs (Mobility Management Entity) for the LTE [24], we do no longer need the time and space switching Matrix. The CSCF / MME acts more as SIP

(Session Initiation protocol) router for IMS and SIP/GTP/DIAMETER router in case of LTE. However, for routing a call, these Network Elements still need to involve directly the Layer 7, which utilize lot of resources and processing power of these Network Elements. This in turn makes the Network Elements expensive and increases the capital and operational cost of the mobile network. As a consequence of the convergence of heterogeneous mobile applications catered by the expanse of the mobile devices preordained for diverse scenarios (P2P, M2M, IVC), the signaling inter-processes between the network and the user equipment becomes more intricate. In that respect, our research work showcases an access topology which emancipates the mobility management process from the application layer which in turn makes the network LITE and future proof. It renders more horsepower for the network to serve the cutting edge applications. It also aids in optimization of the network operations in an endeavor to make it more energy efficient.

2.1. Related work and the issues

The foremost challenge for 4G is to support seamless interoperability [44], handover and roaming for the H2H and M2M devices without compromising the ‘Grade of Service, ‘Round Trip delay’ and latency. It has to meet the desired reliability for the M2M application, while supporting the high throughput demand of a wide variety of H2H multimedia services such as multimedia web browsing, video, online gaming. Various methods [30][31][32][33] for optimising bandwidth have been worked out for mobile and 802.1X networks like dynamic bandwidth allocation according to location / QoS . Some methods [21] adhere to the principle of fixed slot allocation for all cells which is not optimal in terms of bandwidth utilisation given the fact that different device types coexist in the network with different nature of bandwidth requirement. Also, the traffic pattern varies over time and location area. Due to the high proliferation of the M2M devices, there may be significant impacts like network congestions, which will in turn affect the H2H services. The M2M scenarios are characterized by colossal amount of devices that interact frequently or infrequently by small amounts of data. These devices may be clustered in a small zones leading to competition amongst the network nodes. A majority of the devices can initiate connection leading to peaks in signaling and data. This penalizes the non M2M devices.

The architecture of the IMS/LTE networks[23][24][10] is expected to keep large use of IP and use a large spectrum of core and access technologies, stemming from the necessity to cater to various types of mobile applications depending on the device type / capability. Wireless 4G networks create various challenges due to their architectural heterogeneity in terms of differences between access schemes, resource allocation techniques and QoS requirements. To address these challenges, the existing proposals require a significant modification or new Medium Access Control (MAC).

The primary issues that may impact the State-of-the-Art mobile network due to M2M communication are as follows:

- **Radio Network:** The eNodeB in the LTE network needs to connect a large number of M2M devices. The devices will contest to use the same channel leading to collisions.
- **Core Network:** The MME needs to attach a large number of devices. The network also needs to provide the IP and NSAPI address to those devices. A lot of devices imply sporadic and transient use of the bearer, leading to an overhead. The HSS needs to have the subscription and profile definition

for the M2M devices. There can be signaling congestion towards HSS when a large number of devices try to register in the same HSS.

- Unlike a H2H Network, we seldom experience congestion in the data plane for M2M networks. This is because devices send and receive small amounts of data. But a lot of devices transmit and receive data simultaneously leading to congestion mainly in the Enhanced Packet Core (EPC) part between the Serving Gateway (S-GW) [24] and the Packet Gateway (P-GW) [24]. Capacity augmentations need to be done for adjuncts like MME, HSS, AAA servers, CDR processing engine.
- The goal of 4G is to replace the current proliferation of core mobile networks with a single worldwide core network standard, based on IP for control, video, packet data, and voice. The IP protocol used is IPV6, which uses 128 bit addresses vis-a-vis the 32 bit addresses of IPV4 protocol. Hence the IP V6 protocol which is directly used for mobility management in 4G can offer a wider address pool for the mobile users. This is essential to accommodate all the M2M devices in the mobile network. To enhance mobility in IPV6, 'micro-mobility' protocols (such as Hawaii, Cellular IP and Hierarchical Mobile IPv6) have been developed for seamless handovers i.e. handovers that result in minimal handover delay, minimal packet loss, and minimal loss of communication state. However, the core issues of IPV6 to be involved in mobility management still remains.

Four key issues staggering the operational efficiency of the State-of-the-Art mobile network is as follows:

Issue1: Paging support

- The base IPv6 specification does not provide any form of paging support. Hence to maintain connectivity with the backbone infrastructure, the mobile node needs to generate location updates every time it changes its point of attachment, even if it is currently in dormant or standby mode. Excessive signaling caused by mobility leads to a significant wastage of the mobile node's battery power, especially in environments with smaller cell [26]. Therefore, it is impractical to rely completely on location updates, and it is essential to define a flexible paging support in the intra-domain mobility management scheme.
- In order to save battery power consumption of devices, IP paging is proposed as an extension for Mobile IP [25]. Under Mobile IP paging, a device is allowed to enter a power saving idle mode when it is inactive for a period of time. During idle mode, the system approximately knows the location of the mobile node within the paging area comprised of multiple subnets. A device in idle mode does not need to register its location when moving within a paging area. It performs location update only when it changes paging areas. However, with the high penetration of the M2M devices, the size of the paging area will reduce. This will culminate in high signaling network load when the M2M device (say, implemented for a fleet management company) alters the location frequently.

Issue 2: Round trip delays

- According to the IPV6 implementation principle for mobility management [25], the mobile node needs to send binding updates to the home agent in a given periodicity. This action implies that an authentication procedure needs to be actuated for each bind updates resulting in an increase in round trip delays.

- For these reasons a new Mobile IPv6 node, called the Mobility Anchor Point (MAP), has been suggested in RFC 4140. MAP can be located at any level in a hierarchical network of routers. The MAP will limit the number of Mobile IPv6 signaling outside the local domain. As an evolution from IPV6, Hierarchical Mobile IPv6 (or HMIPv6) proposes the implementation of the MAPs. Networks are divided into domains and subnets, with each administrative domain having a Mobility Anchor Point (MAP) at the highest level. Intra domain mobility of a mobile host is handled separately from inter-domain mobility.
- When the Mobile Host (MH) changes points of attachment within the same domain, the MAP of that domain is informed of the change in Care of address of the MH through binding updates. Binding updates are also sent to correspondent hosts within the same domain. This process reduces signaling traffic due to lesser binding updates. Handoff latency also decreases as ‘far-off home agents’ and ‘correspondent hosts’ need not be updated every time a mobile host changes point of attachment. IPV6 based mobility management ensures minimal handoff latency to achieve better QoS for real-time data conveyance. However HMIPv6 has its drawbacks as the hierarchical addressing model may impact the real time applications. Due to the hierarchical nature of the protocol, each anchor point is only aware of the next anchor point down in the hierarchy. Each node stores a mapping of source destination addresses of the previous and next nodes in the hierarchical structure. These addresses are called VCoAs (Virtual Care of Address) [28]. Only the lowest anchor point in the hierarchy stores a mapping of VCoA to PCoA which is the physical care-of-address of the MH in the foreign environment. As the source is aware of only the VCoA of its nearest anchor point, it sets that address in the destination field header of the IP packet. At each hop, the packet is processed, depending on the source address, the new destination address (VCoA) is decided. The packet is then forwarded using the new destination address and the node’s own address as the source address. This processing occurs at each hop and can effect real time applications running at the mobile device as it creates significant delay especially if the CH (Correspondent Host) and the MH are many hops away with the MH part of a big hierarchical structure.

Issue 3: Signaling Overhead

The signaling overhead in GSM for different Access Network related operations is in Table 1. The operations with respect to the below in Table 1 are related to the signaling channel. The mobility management messages in LTE have evolved from GSM/UMTS. In this section we step back to GSM/UMTS to introspect the issue of high signaling overhead in the access network which extrapolates to the core network.

In GSM, have 4 kinds of signaling channel as in Figure 1. Out of these, the Common Control Channel (CCCH) and the Dedicated Control Channel (DCCH) take part in transporting the overhead that is required for these signaling operations. Table 1 shows the overhead required to convey signaling messages related to mobility management and addressing. Not only it consumes bandwidth, but it also entails considerable amount of Layer 7 processing between the handset (Mobile Station) and the Radio Network.

The handset thus consumes battery power to actuate layer 7 functionalities. The Radio Network elements need to have the required processing power to manage all the complex mobility and location management processes for all the active mobile users in the Location Area.

Each operation as depicted in Table 1 is related to a layer 7 functionality, and hence dependent on a handshake process. The lead time to complete an operation directly impacts the time to complete an event, like Call setup, handover, SMS transfer etc. This is indicated in the right most column of table 2 as the acquisition time.

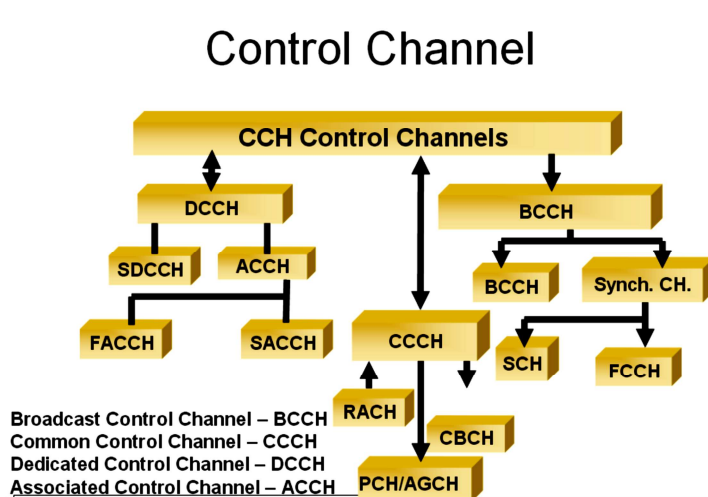


Figure 1 Control Channels in GSM

Service type	MSG length (bytes)		Acquisition time (ms)	
	DCCH	CCCH	DCCH	CCCH
SMS	1180	1000	204.4	133.3
Location update	394	214	81.6	38.6
Call setup	652	472	148.9	88.9
End call	689	500	155.3	93.8
Paging	-	9	-	2.0
Inter-RNC Handoff	-	17	-	2.71
UE offline	199	45	37.7	36.6

Table 1: Signaling message length and network acquisition time for various network operations for users.

Issue 4: Impact of Network performance and optimisation on cell planning and frequency reuse

Much of the network utilisation depends on how cells are oriented in the network; the frequencies that are used in the cells and the frequency re-use factors. The frequency reuse factor is the rate at which the same frequency can be used in the network. It is $1/K$ where K is the number of cells which cannot use the same frequencies for transmission. Common values for the frequency reuse factor are $1/3$, $1/4$, $1/7$, $1/9$ and $1/12$ (or 3, 4, 7, 9 and 12 depending on notation).

In case of N sector antennas on the same base station site each with different direction, the base station site can serve N different sectors. N is typically 3. A reuse pattern of N/K denotes N sector antennas per

site. Common reuse patterns are 3/3, 3/9 and 3/12. Figure 2 depicts the frequency reuse pattern within a cluster of cells.

If the total available bandwidth is B , each cell can only utilize a number of frequency channels corresponding to a bandwidth of B/K , and each base station site can use a bandwidth of BN/K .

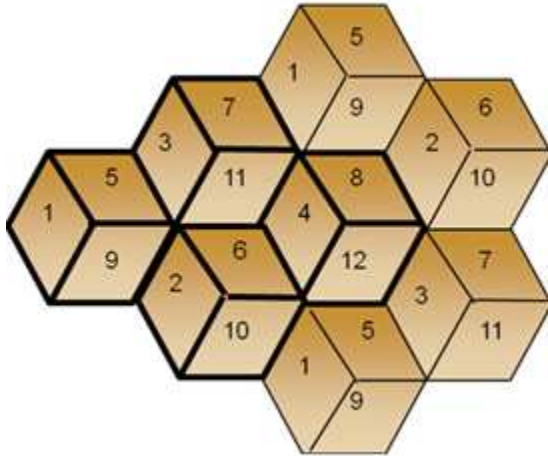


Figure 2: Frequency Reuse within 3 sectors of the hexagonal cells

Hence, the frequency reuse factor is directly related to cell optimization and spectral efficiency.

The cell planning and frequency re-use is done also taking in mind that a good C/I ratio is achieved.

An area to be supplied with radio service is divided into cells. Each of these cells is assigned a range of frequencies ($f_1 - f_6$) and they each have corresponding radio base stations. The group of frequencies can be reused in other cells bearing in mind that it is not reused in adjacent cells as that will cause co-channel interference. Co-channel interference results when frequencies are reused in adjacent neighbouring cells. A reuse distance, D is calculated as :

$$D = R\sqrt{3.N}$$

where R is the cell radius and N is the number of cells per cluster.

Cells may vary in radius in the ranges (1 km to 30 km).

Frequency hopping is another technique that can increase the spectral efficiency, and the C/ I ratio. But it is applicable only for the traffic channel, and not for signalling. 4G Networks however attempt to attain Frequency Reuse Factor of 1 [26], by implementing Dynamic Radio Resource Management. New generation 4G systems like LTE and WIMAX are designed to support frequency reuse-1 mechanisms to enable a universal frequency reuse pattern providing operators with best achievable use of their valuable spectrum. Ever increasing interest is shown by companies and operators to find mechanisms to allow reuse1 deployments. Of late, new technologies like Inter Cell Interference Coordination (ICIC) and their application will facilitate universal use of Reuse-1 networks.

ICIC eliminates signal interference at cell-edge areas covered by base stations of the 4G (LTE) mobile phone system. The new technology is able to autonomously allocate frequency bands in order to reduce radio-wave interference in accordance with user distribution around cell-edge areas of overlapping cells of each adjacent base station. By deploying this technology to base stations, it is possible to double throughput at cell-edge areas, which would otherwise experience reduced transmission speeds due to signal interference.

OFDMA based 4G networks can support reuse one deployment through the use of;

- Ability to power control and vary the coding rate of control channel;
- Fractional power control with coordination with controller based overload control messages in UL;
- Support of very low code rates;
- DL and UL ICIC further enhance single cell frequency reuse;

One of the key goals is to increase spectral efficiency and overall SINR of the system. However, users at the cell edge are particularly susceptible to increased interference resulting in reduced throughput due to higher transmit powers required and inter-cell interference. Therefore interference coordination as methods to reduce inter cell interference is gaining momentum and industry attention. Most of the above mentioned factors and tasks are dependent on layer 3/5/7 signalling activities. This not only consumes bandwidth and increases cost, but necessitates the processing of complex operation within the network devices. The network devices need to interact and interoperate for all these signalling activities, which mandates the networks elements to be intelligent and fast in processing for attaining a defined QOS.

The inability of the radio network elements to efficiently handle these operations will result in call failures, call drops, high call setup time, lower ASR (Answer Supervision Rate) and offered network quality.

2.2. SoftRAN: Software Defined Radio Access Network

A significant endeavour to encounter the challenges pertaining to Inter-cell interference, reduced data rate at cell edges, frequency reuse, cell planning, small cell deployment is witnessed in the concept of SoftRAN. In the state-of-the-art mobile networks like 3G or LTE, the radio network comprises of base stations acting independently. In LTE network, the eNodeBs engage themselves in cooperative communication and distributed coordination mechanisms such as SON (self-organizing networks) over the X1 interfaces. But they are individual entities when interfacing with the MMEs (Mobility Management Entities). Smaller the cells served by the eNodeB, higher are the coordinated control plane decisions amongst neighbouring cells and with the MMEs. The issues that crop up are high bandwidth requirement due to increased signalling overhead, imbalance in traffic and higher complexity of interface management. Further, distributed coordination algorithms tend to become more complex, since often they require iterative and periodic adjustment of radio layer allocation decisions that are hard to get right at scale.

Software defined RAN [60] logically aggregates all base stations deployed within a geographical area and treats it as a virtual big-base station with a three dimensional resource grid. Instead of treating a base station as an individual entity with its own set of resources, a virtual base station is abstracted as a three dimensional grid of space, time and frequency slots and code them as a soft radio access control and data plane. APIs are defined between the radio elements and the control plane which allow radio elements to

update the global view at the control plane and also allow the control plane to communicate radio resource management decisions back to the radio elements.

However, a single base station abstraction cannot be achieved without changes to LTE standards. The client will continue to sense multiple base stations and at each handover. The eNodes in vicinity of the eNodeB initiating the handover will coordinate with each other to actuate a smooth handover. However with SoftRAN, it is anticipated that the handover mechanism will require less resources due to global cognizance of the channel conditions at various locations in the coverage area. This implies smoother handover, lesser call drops and diminishing chances of ‘back and forth’ handovers between the same pair of base stations. With a smoother control on transmit powers across multiple base stations and better management of interference, the clients would also experience a more stable connection. Conceptually, SoftRAN is analogous to traditional SDN with a centralised RAN architecture. This logically centralized entity then maintains a global network view and provides a framework on which control algorithms can be implemented in a modular fashion.

2.3. Network Interoperability issues with non-human device types

2.3.1. Machine to Machine

The concept of Machine-to-Machine (M2M) communications fits into the new trend of devices that we see around us disseminating information through the mobile network to the remote peer or to the cloud. These objects have their own Internet Protocol addresses. They are embedded in complex systems and interfaced with sensors to obtain information from their environment (e.g. food products that record the temperature along the supply chain).

Some guidelines are given for particular aspects of this M2M technology in report such as ETSI’s TS 102.689 [29]. But a unified approach contemplating the coexistence and seamless interoperability between the different device types is still ‘work in progress’. This poses a risk, as the State-of-the-Art mobile networks were designed keeping in mind Human to Human (H2H) communication. Our contention is that the State-of-the-Art mobile network may not be resilient to the data surge likely to be caused by M2M applications. It is probable that a group of M2M devices for a particular industry application may initiate services and access the mobile network at the same time, causing network congestion and degrading the quality of other services. For example, a fleet management company may have moving carriers in different geographical locations trying to access the network and upload data at the same instance. Considering the high proliferation of these devices, an avalanche effect on signaling and data traffic can be generated, thus impacting the gross QoS (Quality of Service) offered by the network. It is imperative that the mobility management processes need to be revisited in order to cope up with the surge in the mobility management messages in the mobile network and to maintain the same QoS without being heavy on the bandwidth. Attempt should be made to reduce the interaction with the network and to simplify the mobility and location management related processes.

Challenges faced by the present mobile network in the light of the M2M scenarios are:

- Number of M2M devices can be fairly large compared to the H2H devices. So the length of the addressing parameters that are used today (MSISDN, IMSI or even the IPV6 addresses) seem to be insufficient to identify the high volume of devices in the network.
- Power management will play an important role. For example, a M2M device meant for location based application in a freight container will need to sustain for weeks, if not months, may be without an external constant power source. So the battery power should be carefully conserved, and one of the effective ways to do so is to reduce the signaling interaction between the handset and the network to maintain location and presence information.
- Variation of QoS requirements: Some real time applications have high demands on latency and reliability (as for example, accident sensors). While the other applications like reporting services may be contended with a lesser priority scheme but with a higher bandwidth requirement.
- Different traffic patterns: Some M2M devices may not transmit any data for months, for example a crash sensor, while other sensors may transmit data continuously or periodically.

2.3.2. Intelligent Vehicular networks

In general vehicular networks have been devised with the concept of vehicle to vehicle/ control center, or vehicle to roadside communication. The mobile equipment in the vehicle equipped to cater to specific applications are called On Board Units (OBU). The vehicles can also communicate with other vehicles or central office for voice or data communication. They can interoperate with the roadside equipments placed at a given intervals, and are called Road Side Units (RSU). The RSUs can act as base stations or access points depending upon the modulation technology adopted for the vehicular network.

Application scenarios for vehicular communications can be roughly grouped into safety (e.g. collision warning, roadwork warning) and non-safety (e.g. voice communication, media, Internet access). These application types put different and partially conflicting requirements on the system design. Non-safety applications related to entertainment, information and content has a continuous pattern of data conveyance over a stable data channel which can reliably transceive data throughout a vast geography. Ease of extending coverage, latency and jitter play an important role in determining the Data and service quality.

In contrast, safety applications data is normally disseminated sporadically on need basis. They may not require a very high bandwidth. But the data transfer mechanism should be fail safe and reliable.

Vehicular communication aims to serve the following application scenarios.

- Presence of obstacles on road.

- Group cooperative driving
- Real time monitoring on engine and transmission operating parameters
- Emergency braking of a preceding vehicle.
- Information about blind crossing, school proximity, railway crossing etc...
- Entries to highways.
- Geo location and tracking applications
- High speed internet access.
- Entertainment media applications
- Electronic toll collection.
- Parking space locator in cities.
- Nearest petrol pump, breakdown assistance

The primary challenges of Vehicular networks area :

- Scalability: Shared Medium \Rightarrow limited resources & capacity
- Robust and reliable communication even at high speeds
- Cope with sporadic data bursts from specific location areas
- Efficient data dissemination schemes for large number of vehicles within a short span of area, like in traffic jams

2.3.3.VANET (Vehicle Area Networks) implementing IEEE 802.11p

In a typical implementation of VANETs, the network is readily accessible via WiFi, DSRC, WiMAX, 3G, LTE, etc [43]. In the introduction we brought out the concerns related to use of traditional 3G/LTE networks for use in vehicular networks, as it may induce congestion in those networks thus impacting the service for all users including Peer2Peer. But it renders an important capability to the vehicular network, which is mobility.

European Car-to-Car Communication Consortium (<http://www.car-to-car.org/>) is involved in the standardization process of the IEEE 802.11p automotive communication standard for VANETs. Noticeably, it is comparable to the DSRC (Dedicated Short Range Communication) technologies coined by USA. Both standards use a communication frequency band around 5.9 GHz and rely on the OFDM modulation scheme. The WLAN-based technology proved to be usable for the general task of exchanging messages between vehicles or RSUs in an ad hoc manner. However, the problems arise when there are applications which require structured and unperturbed communication between the device and the network (eg, the RSUs) which proves to be difficult. Moreover, WLAN has issues of seamless mobility management and roaming which is still work in progress. There is still no fool proof solution that can provide global roaming, network omnipresence, yet does not impact the performance of the heterogeneous mobile device types and applications

2.3.4.IVC using DOLPHIN

Inter Vehicle Communication (IVC) is realised by engaging the car in communication process and electing it to act as the base station. But this process is hindered when the car acting as the base station departs from the group. For this reason, in IVC systems it is necessary to use the autonomous dispersion type of network formation technology in which the existence of a base station is not hypothesized. In addition, the communications data in the IVC system is driving control data, for actuating “coordinated driving,” so it must be of high quality and capable of real time transfer. The DOLPHIN (Dedicated Omni-purpose inter-vehicle communication Linkage Protocol for Highway automation) [42] communication protocol for use with IVC systems has been developed by OKI Japan. They have adopted the Carrier Sense Multiple Access (CSMA) method which is well suited as a wireless access method for building autonomous dispersion type networks and which does not require a base station. Using received signal strength information (RSSI), carrier sense determination is made, and transmission processing of the physical frame control part is actuated. When a signal is not sent on time as a result of the carrier sense, the process is re-performed after a random period of time passes. However, no mobility management schemes have been considered while designing this system which is the primary drawback of the solution.

2.4. Vehicular Closed User Group network essentials

The typical radio trunking networks are single tier broadcast type . The main network elements are the central transceiver, channel switching equipment at the central station and the radio trunking equipment plugged in the vehicle. There are two types of radio channels: control (or data) and traffic (or voice) channels. There is a single control channel and the rest are traffic channels. The control channel is used to send digital information between the radio units and the computer equipment controlling the operation of the system. Generally the control information is specific to the manufacturer

The radio unit continually listens to the control channel waiting for instructions. When a call is invoked the transceiver invokes a digital message over the control channel and informs the central office that it needs a channel to set up the call .The central office equipment accepts the request for a channel and assigns a free channel by return digital message over the control channel. The radio unit receives the working channel assignment and fixes it's transmit and receive frequencies to the new channel. Following this, voice communication can commence through the traffic channel. The disadvantage of radio trunking systems is limited channel capacity and the maximum number of users that can be served simultaneously. It can only reach a limited number of users. Hence for a large car fleet group, or Railways, Hospitals, Factories, Cruise ships radio trunking systems are generally not conceded. For a relatively large sized network active network elements like Mobile Switches, base stations are required for mobility management .Many of these devices operating in the network space along with the H2H devices increases the complexity and costs.

2.5. Nano Networks

In these nano devices, we see nano scale fractal antennas for supporting multiple spectra and broadband. Bell Lab researchers are focusing on building arrays of carbon nano-tubes grown on silicon plates which could serve as high-efficiency electron-emitting vacuum tubes. The Nano mobile phones have nano scale

power storage devices based on fuel cells. They will also derive power from natural sources like solar energy, but not with conventional solar cells. Nano grass [44] provides a major boost in efficiency to harness solar power in the new generation mobile devices.

The nano mobile device faces challenges to operate in a mobile network, due to limited power resource and restrictions in RF power output. Moreover, the RF spectrum needs to be perpetually scanned in real-time with low power consumption. Significant amount of processing speed and power is needed to run the algorithms which enable intelligent use of spectrum, gathering channel state information and fast adaptation to dynamically changing radio environment.

2.6. Motivation and objectives

The legacy and LTE networks are too closely knit together considering the basic networks processes that leads to intricacy and overhead. While the present generation focusses mainly on promising towering bandwidth per user, we feel that substantial emphasis has not been imparted on a greater network optimisation. Hence, there is a room for improvement. Some major drivers to define the next generation mobile technology will be, for example, to attain lower battery consumption, lower outage probability (better coverage), high bit rates in larger portions of the coverage area, cheaper or no traffic fees due to low infrastructure deployment costs, or higher aggregate capacity for many simultaneous users (i.e. higher system level spectral efficiency).

We aim to achieve this by proposing the Novel Multiple Access methodology. Some of the benefits that we will derive from this are as follows.

At a high level, the proposed technology has manifold benefits

1. The access layer does not need intelligent network devices like BTS/BSC which helps in forming the cellular network area and actuate the location and call management.
2. So faster and cheaper implementation can be achieved.
3. Reduction in mobility management functions moderates the overhead required for signalling activities.
4. Absence of cellular architecture for Closed User Networks wards off the necessity of cell planning and management. Peer to peer communication does not involve any intelligent core or access network elements.
5. For a global roaming network, the cellular architecture is followed. However, it is simpler in design compared to the existing network topologies, in terms of location management and handoff.

Summing up, this reduces the cost of Setting up a Mobile network plus the cost of operations and maintenance of the network. As both the access and the core network are more simplified, hence the cost of cell planning and maintenance is much reduced for a global mobile network. The radio and the core network elements need to be less intelligent as the Mobility Management part is drastically simplified. The signalling overhead and signalling operations between the User Equipment and the Core Network is reduced, implying less time for call set up, more battery power and simplified UE and the RAN (Radio Access network) / CN(Core Network). Network Efficiency and Quality depends upon various factors

related to Mobility management, Location management, addressing, handover network cell design and frequency allocation. The RRM (Radio Resource Management) implemented in GSM /UMTS/ LTE [11] is explained below briefly. Our research work proposes an alternative topology primarily attempting to simplify these processes.

2.7. Objectives and Synopsis of the proposal

In a traditional multiple access scheme, the physical plane is not involved in identification of the specific users whose data symbol it bears. User identification is predominantly a task attributed to higher layers. Addressing a mobile node in a cellular network proves to be a convoluted process of location and mobility management encompassing layer 1 to layer 7 of OSI. These processes deplete network resources and power. Our research is intended to streamline these processes and to set out an architecture which disassociates the layer 7 from the ‘addressing’ and ‘mobility management’ processes. This will significantly alleviate the network from the incessant control signalling between the mobile device and the network.

Unlike the traditional multiple access schemes, with this proposed method, each mobile node will be directly identified in the complex plane. A mobile node is represented by a fixed symbol with definite spatial coordinates in the complex plane which has a total of 8 symbols in the outer ring. Hence 8 symbols will represent 8 users in the complex plane.

We use circular 8PSK/ QPSK with two concentric circles; the outer circle having 8 symbols on 8PSK and the inner circle having 4 symbols on QPSK. There will NOT be a dedicated channel for carrying the signalling information. The main Channel is subdivided in multiple channels, each of which will have 8 symbols per frequency band for addressing the user and 4 symbols for user data. The rate of transfer of the symbols in the outer circle will pertain to the call attempt rate, while the rate of transfer of the symbols in the inner circle will pertain to the actual traffic data rate.

We propose a smart multiple access topology which will be able to perform specific layer 5 and 7 processes related to addressing and identification of a node in the RAN, right at the physical layer. After the addressing part is accomplished, and the data channel (say a time slot for the traffic conveyance) is identified, the data communication for the supplementary services will follow the conventional methodology of a GSM/UMTS /LTE network.

3

Smart Mobile Network Access topology

2.8. Introduction

In this chapter, we present the basic philosophy of the proposed smart access network technology [2] and how it fits within given boundaries of the core network topology. We will discuss in details how the symbols are used to convey information for the purpose of addressing and also the user traffic. The basic addressing mechanism is described, the high level call flows are discussed and then the internal architecture of the mobile station is presented.

As we highlight the details of the timing frame structure and the associated calculations, we will also present the Simulink simulation model to validate the concept of physical layer addressing. In the subsequent section, we focus on the type of the mobile network, say whether a Personal Communication System (PCS) network, or a closed user type of network, or a Global Mobile Network. We mainly discuss the modulation scheme, the timing diagrams and high level schematic of the mobile device w.r.t each network implementation. In a small scale closed user type of network, it will be possible for node to node communication without the involvement of any intelligent core network equipment. In this chapter we mainly focus on closed user networks and small scale mobile networks categorized under Personal Communication System. The primary components which comprise the network are discussed. In this chapter we coin a new Network element called the coordination processor which is an analogue of Mobile Switching Centre of 2G/3G network, CSCF for an IMS network or MME (Mobility Management Entity) for a LTE network.

2.9. Novel multiple access scheme

The main features of the smart multiple access process is as follows.

- 1) It actuates layer 5 and 7 processes related to addressing and identification of a node in the RAN, right at the physical layer [1][13]. Each node is assigned a fixed symbol coordinate in the physical plane, a time slot in the time frame and a physical channel. The symbol pertaining to the specific time slot is used to address a user. After the addressing part is accomplished, the time slot is seized for traffic data conveyance.
- 2) The time/space switching matrix of a 2G network, CSCF (Call Session Control Function) of IMS network [10][23], MME (Mobility Management Entity) of LTE Networks [24] is substituted by a Network Element termed as coordination processor. It is responsible for the addressing and mobility management functionalities employing the physical layer. It also takes care of some Layer 7

functionalities that are still required for AAA (Authentication, Authorisation and Accounting), supplementary services and Value Added Services. The access layer also has reduced addressing and mobility management functionalities, and hence requires less intelligent devices which otherwise would have required to support extensive signaling inter operations.

- 3) The RAN (Radio Access Network) will comprise of ad hoc repeaters, referred to as Access Points in the network area. They reside at the same network hierarchy level as the coordination processor of the core network. Unlike BSS (Base Station System) of RAN in GSM/UMTS, where the access network devices are identified by specific global addresses (in E.212 or E.164 Numbering plan format), the ad hoc repeaters in purview of the smart network topology will not have any specific network addresses except the Geographical Information System (GIS) addresses.

The detailed description of the multiple access technology and the frame structures is as follows.

We assume a TDMA/FDD multiple access. The available bandwidth is divided in multiple sub frequency bands. Alike GSM, the frequency sub bands are called AFCRN (Absolute Radio Frequency Channel Number). The AFCRN are numbered in numerical order. The number of AFCRN depends on the bandwidth that is required for the mobile network. Each AFCRN is linked to one Time Frame. Fig. 3 shows one TDMA frame, which includes $Tn + 1$ time slots (numbered from $T0$ to Tn). The data rate required for the traffic burst is denoted by f_d . f_p denotes the sampling rates of the time slots before it is used to convey the traffic data. f_r denotes the refresh frequency for the given frame and t_r

$$f_d \gg f_p \gg f_r$$

The modulation scheme implemented is a blend of M1 PSK (for outer ring) and M2 PSK (for inner ring), where $M1 > M2$. Fig. 4 shows the proposed constellation in the case of $M1=8$ and $M2=4$. The constellation diagram in Fig.2 pertains to a specific time slot, $T1$, of the frame. The frame corresponds to a specific AFCRN. Symbols of the outer ring are conveyed at a rate of f_r and the symbols of the inner ring are exchanged at a much higher rate, f_d

In particular, we could set $f_d = 8 \times f_r$. The inner ring comprises of M2 symbols for user traffic, and the outer ring comprises of M1 symbols that will be used for the purpose of addressing the users. In the proposed multiple access scheme, a user is identified in the network with respect to the symbol coordinate in the complex plane. The 8PSK symbols are continuously rotated with $3\pi/8$ radians per symbol. The rotated symbols are defined in equation 1.

$$\hat{S}_i = S_i \cdot e^{j i 3\pi / 8}, \text{ for } i = 1 \dots n \quad (1)$$

Where s represents the symbol coordinates

There is no separate time slot for conveying the control signaling and the data part. A single time slot can carry M1 symbols, used for addressing M1 users and later, the same time slot will be used to convey the symbols for data traffic employing M2 symbols of the inner ring of the complex plane.

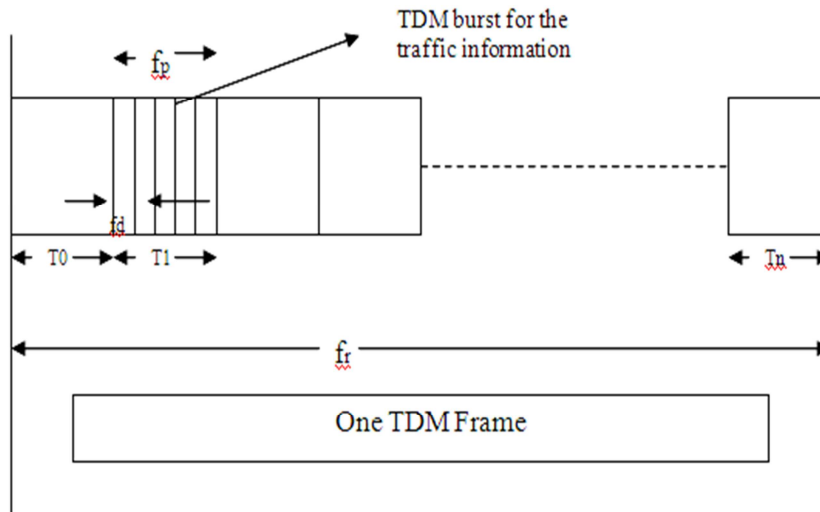


Figure 3: TDMA frame

Therefore, M_1 users, which have been allocated M_1 different symbols of the outer ring could use the same time slot and frequency sub-bands.

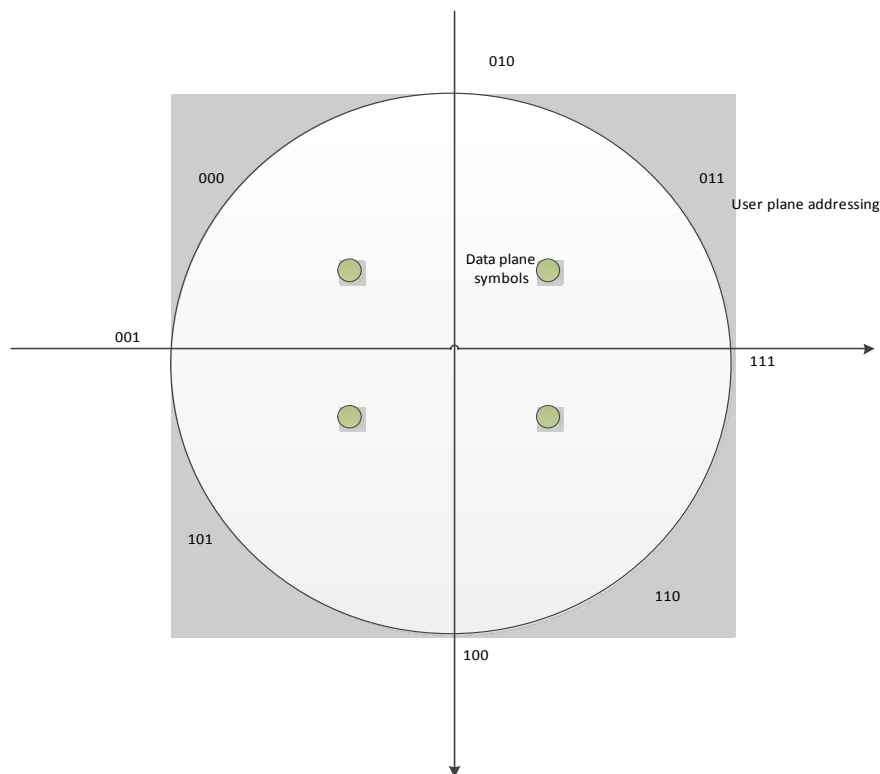


Figure 4: Constellation Diagram

Therefore, even if a primary frequency is allocated to a given user, it may happen that the particular frequency is in use by another user which has been assigned the same time slot and frequency, but a different symbol in the complex plane.

According to the presented scheme, we need to specify:

- 1) The action to be undertaken when in a given frequency sub-band, the assigned time slot is already in use by one of the other M1 users, characterized by different symbol coordinate.
- 2) How the frequency sub-band, the time slot and symbol coordinate for addressing the user in the network are assigned to the user during the provisioning process.

These two points will be explained in the following two paragraphs 3.3 and 3.4.

2.10. Frequency scanning

As there are 8 users that can be allocated the same time slot (as in the above section), hence it can happen that multiple users allocated the same time slot and primary AFCRN attempt to invoke some messages simultaneously. Such a scenario will lead to collision in the network. Hence when the mobile user detects that its time slot is already in use for traffic/ signaling activity, it attempts to find out another timeslot exactly in the same position in the time frame but associated with another AFCRN.

The AFCRN scanning algorithms are programmed within the user equipment, as well as the Coordination processor of the network (discussed in the subsequent section).

If in a given frequency sub-band, the assigned time slot is already in use by one of the other M1 users, characterized by different symbol coordinate, the Mobile Station (MS) or the network may allocate dynamically another primary frequency from a different frequency band to that user. Fig. 2 shows the processes involved for channel acquisition [8][11].

The network will apply any one of the sequence for paging the MS at random. The target MS listens to the time slot and the particular symbol assigned to itself in all the frequencies. So it is not important which sequence is followed to find the first frequency in which the specific timeslot pertaining to the target mobile is free.

The handset and the network uses proactive channel scan algorithms mainly based on one of the following schemes chosen device.

- Alternate Even
- Alternate Odd
- LRU (Least Recently Used)
- Most Idle
- Round Robin Clockwise
- Round Robin Counter Clockwise
- Sequential Bottom Up
- Sequential Top Down

This implies that if a user has initiated a call towards one of the M1 users pertaining to different symbols in the same time slot which is already busy due to use by another user, then the MS or Network (according to the direction of the call) will hunt for that particular time slot in the next frequency band according to the scanning algorithm until it finds one.

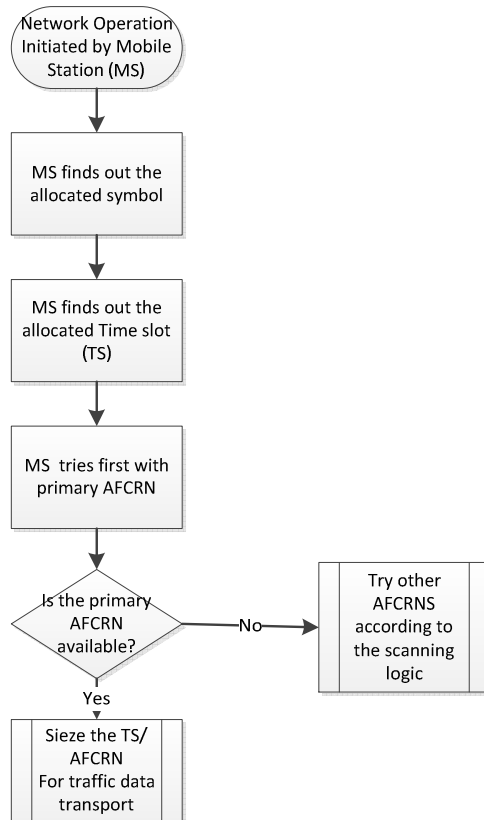


Figure 5: Sequence of the processes to hunt for AFCRN and seize the time slot for data transport

During a mobile terminating call, the network can apply any one of the scan sequences as in Fig 3 to find an unused AFCRN for paging the MS at random. The target MS listens to the time slot and the particular symbol assigned to itself in all the 15 frequencies. Just alike LTE/UMTS, the scanning algorithms should be handpicked by the device modem manufacturer. The frequency scan also considers the strongest pilot based on parameters like RSSI (Received Signal Strength Indicator). There is no specification as such defining these scanning algorithms, and they are proprietary to the chipset vendors, like Qualcomm. Hence, in this study we will not focus on the technicalities of the frequency scanning algorithm and how it can be rendered in the hardware as a firmware.

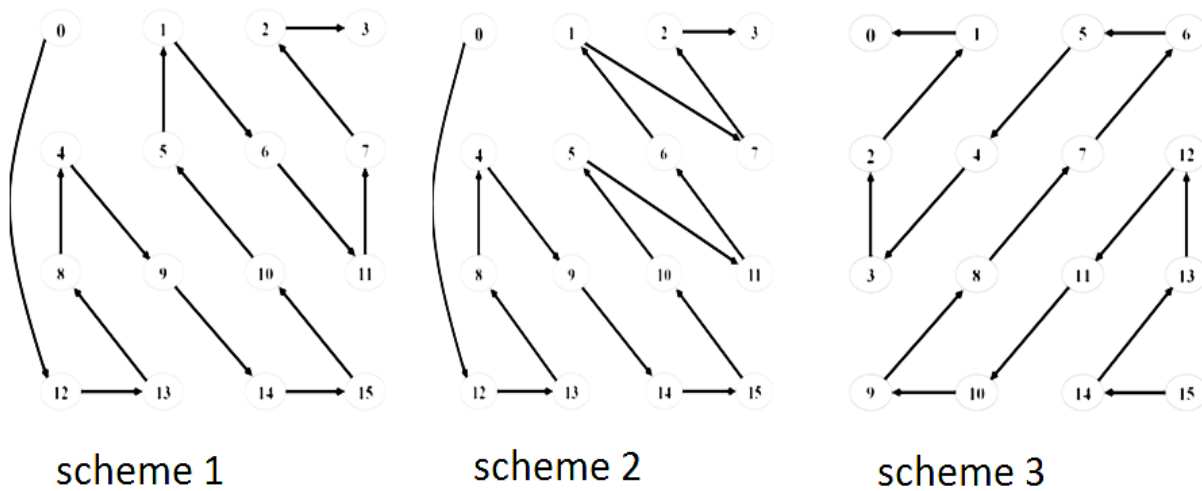


Figure 6: Frequency scanning algorithm

Hence it can be inferred that if a user has initiated a call towards one of the M1 users pertaining to different symbols in the same time slot which is already busy by another user, then the MS or Network (according to the direction of the call) will hunt for that particular time slot in the next available frequency bands according to the scanning algorithm until it finds one. The MS to be paged at the other end (by the network) acts as a silent listener and scans for the specific symbol coordinate and time that has been allocated to itself to search for a page signal.

In comparison, the frequency and cell scan process for UMTS/LTE is described below at high level.

When UE boots first time in a radio network it doesn't have any information of the AFCRN. It has a default search mechanism for the AFCRNs. The UE shall scan all RF channels in the available bands to find a suitable cell. The UE searches for the cell with the strongest pilot frequency.

After the first time successful network attach had happened, the USIM (Universal Subscriber Interface Module) stores the cell information as well as the carrier frequency.

During the next time when the UE tries a cell selection, it uses the stored information of carrier frequencies and optionally also information on cell parameters, from previously received measurement control information elements or from previously detected cells. Once the UE has found a suitable cell the UE shall select it. If no suitable cell is found the Initial Cell Selection procedure shall start. The UE measures the RSSI (Received Signal Strength Indicator) for each channel, and creates a list of each channel numbers with the measured RSSI.

2.11. Address of mobile node as a function of time frame, time slot and symbol

The address of a Mobile Station (say the E.164 Format or the MSISDN in case of a GSM network) is a function of Time slot, frequency, symbol coordinate [1][13]. The numbering plan format of this address is however dependent on the type of network topology. For a full scale Global Roaming Network, this address is IMSI (International Mobile Subscriber Identity), with a E.212 Numbering plan instead of a MSISDN (E.164). Therefore, in such a case there will be a E.164 address of the handset (used by the user to make and receive calls and SMS) which will be linked to the E.212 address (IMSI), both of which will be defined in the subscriber profile of the HSS (Home Subscription system) / HLR (Home Location Register).

Due to the fact that we have layer 1 addressing, the system needs to formulate directly the layer 1 related parameters assigned to the handset right from the node address. As discussed earlier that for a global roaming network which is the scope of this paper, this address will be an E.212 address or IMSI. The mobile station will have the necessary algorithm to generate the layer 1 parameters from its own IMSI from the given logic as discussed in the following section. This is required when the mobile station attempts to originate a call.

Similarly during a terminating call scenario, the coordination processor at the core network will also be able to derive the Layer 1 parameters (which are the symbol coordinate of the user, time slot and the primary frequency number) and generate a page message for the B number for the termination leg. The methodology indicating the procedure to derive the layer 1 parameters from the address of the user is provided below.

Say, there are a total of n time slots in the TDM frame per channel (frequency sub band). Let there be m frequency bands. Let us assume that 8 users exist per time slot. So the total number of users that can provisioned / defined are: $m \times n \times 8$

We could reserve:

- 1 digit for number of users per time slot, the range of value can be 1 to 8.
- 4 digits for max number of time slots per frame
- 3 digits for number of sub bands, each band for one time frame.
- 2 digits (dynamic) used to depict the frequency difference, always on the positive side. The generation of the last 2 digits is dependent on the leg of the call.

Hence the address of the node can be allocated by

Address of the node = $\leftarrow 5 \text{ digits for } n \rightarrow \leftarrow 1 \text{ digits for symbol} \rightarrow \leftarrow 3 \text{ digits for } m \rightarrow$

Hence when a mobile node attempts to set up a call to another node, it checks the number and interprets the Time Slot (TS), symbol number and also the associated frequency with which it invokes a layer 7 message towards the network. In case the particular time slot in the frequency band is busy for a user represented by another symbol in the same time slot, the frequency scanning commences as per the schemes in Fig 3.

2.11.1. Flexibility of system bandwidth

All the existing access technologies for mobile networks have a fixed spectrum bandwidth requirement. Even for a network that is smaller in size with a lesser customer base, they still need to invest in the same minimum spectrum bandwidth.

With our smart network approach, the bandwidth requirement of the physical layer is directly dependent call attempts made in a network. Hence it is directly dependent on the customer base size. So smaller networks will be able to optimize the spectrum requirement accordingly, which is seemingly difficult with all the existing technologies. This is discussed in the following section.

The number of time slots in the TDM frame (pertaining to a frequency sub band) and the scanning frequency of the time frame primarily depends on the factors below.

1. Call attempts per second of the network and the data rate offered per time slot when the time slot is used for traffic for a particular user.
2. It also depends on the maximum number of users that the network can address at a given point of time.

Initially, a time slot in a time frame (for a specific AFCRN) is used for addressing a mobile node. Let the time slot be denoted by ' t_p '.

Let us denote with ' t_d ' the data burst for user traffic

' t_p ' is sub divided in multiple time slots (8 time slots) for data transfer.

Let ' t_r ' be the time duration of the time frame pertaining to a AFCRN allotted for a mobile user.

Hence,

$$t_r \gg t_p \gg t_d$$

Let us say there are 'm' frequency bands (AFCRN)

So the Total available time slots are = $m \times n$

Say t_r is the time required per frame.

$$t_r = 1 / f_r$$

$m \times n$ call attempts can be made in t_r time period

Hence Call Attempts per second = $m \times n / t_r$

Therefore, the permissible call attempts per second can be expressed as:

$$A = m \times n \times f_r$$

As shown in Fig 7, the data rate will be bursty in nature, which means that the data for a given user will arrive at a frequency of f_r and followed by a gap of time period

$$t_g = t_r \times (1 - 1/n) = t_r \times (n-1)/n \quad (2)$$

(where t_g = time gap between 2 TDM bursts for a given time slot for data transfer)

Note: for large n,

$$(n-1)/n \approx 1, \text{ so } t_g = t_r$$

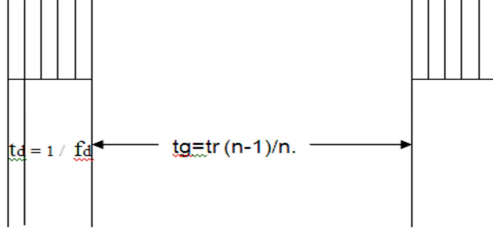


Figure 7: Time Gap between the bursts

Let $t_{g \max}$ be the maximum time gap allowed between 2 consecutive TDM bursts for carrying the data for a given user.

Let there be k number of time slots for the data transfer within the time slot allocated for addressing

$$\text{So, } t_n = k \times t_d$$

T_0 to T_n makes one frame. Let us say there are 'm' frequency bands

So the total available time slots are = $m \times n$. Say t_r is the time required per frame

$$t_r = 1/f_r$$

So $m \times n$ call attempts can be made in t_r time period

Hence Call Attempts per second (A) = $m \times n / t_r$

$$\text{Now, } t_g = t_r (n-1)/n \quad (3)$$

$$t_g = t_r [(A/m.f_r) - 1] / (A/m.f_r) = k \times t_d [(A/m.f_r) - 1] / (A/m.f_r) \quad (4)$$

$$t_{g_{\max}} = k \times t_d [(A/m_{\max}.f_r) - 1] / (A/m_{\max}.f_r) = k \times t_d [1 - m_{\max}.f_r/A] \quad (5)$$

The total bandwidth required for the network will be dependent on the Busy Hour Call Attempt encountered by the network. As the physical layer is directly involved in identifying and addressing the user, hence the total bandwidth required per network will depend on the number of simultaneous users making or breaking calls in the network.

As the addressing and mobility management has much less dependence on layer 7, the total bandwidth requirement and the number of sub frequency bands realized does not have a strict boundary unlike GSM which is defined in the specifications. This flexibility is a conceivable advantage compared to state-of-the-art networks which require fixed spectrum bandwidth allocation for operations.

The values which are proposed for the parameters used in Equation 3 through Equation 5, for a mid-sized network with no frequency reuse (one cell architecture) are as follows. The values are proposed according to the GSM 5.05x specifications in an attempt to keep the same frame duration and structure.

$t_d = 1$ symbol @ rate of 270833 symbols/sec and $3.692 \mu s$ duration for voice /data. The symbols pertain to the inner circle of the constellation diagram.

Hence $f_d = 1/t_d = 1/3.7 \mu s$

$t_r = 8$ symbols (referred to as Burst) .

t_r is a single time slot carrying symbols of the outer ring on APSK. After the traffic channel is established, the t_d shifts to traffic mode and then starts to carry the traffic data at frequency of f_d .

$t_r = 156 * t_d$

$t_r = 577 \mu s$

Considering a 26 Multi Frame arrangement, the total number of Time slot (t_g) that can be realised per sub channel frequency is $26 * 8 = 208$.

Max number of simultaneous users = $208 * \text{Number of AFCRN} (124) = 25792$.

To ward off inter symbol interference, the maximum bit rate is 271 kbps per user.

For a mid-sized network, we do not follow the cellular methodology, and hence there is no re-use of the frequencies in the network area.

This implies that reuse factor is 1.

Number of AFCRN for the Location Area (there is only one big cell) = 124.

2.12. User Identification and call establishment between the mobile nodes employing physical layer addressing.

As depicted in the call flow in Fig 8, the mobile node can generate a symbol positioned in the complex plane according to the called address. This is true for a Closed User group network where the addressing scheme is known to the device itself. For a full scale mobile network, it is an active component in the core network equipment which takes care of the user identification. The symbol is placed in an appropriate time slot allocated for the device and encapsulated in the time frame. The frequency channel is chosen according to the frequency scanning process defined in section 2.3.

The call flow in Fig 8 demonstrates how mobile stations can establish semi-permanent connection between themselves.

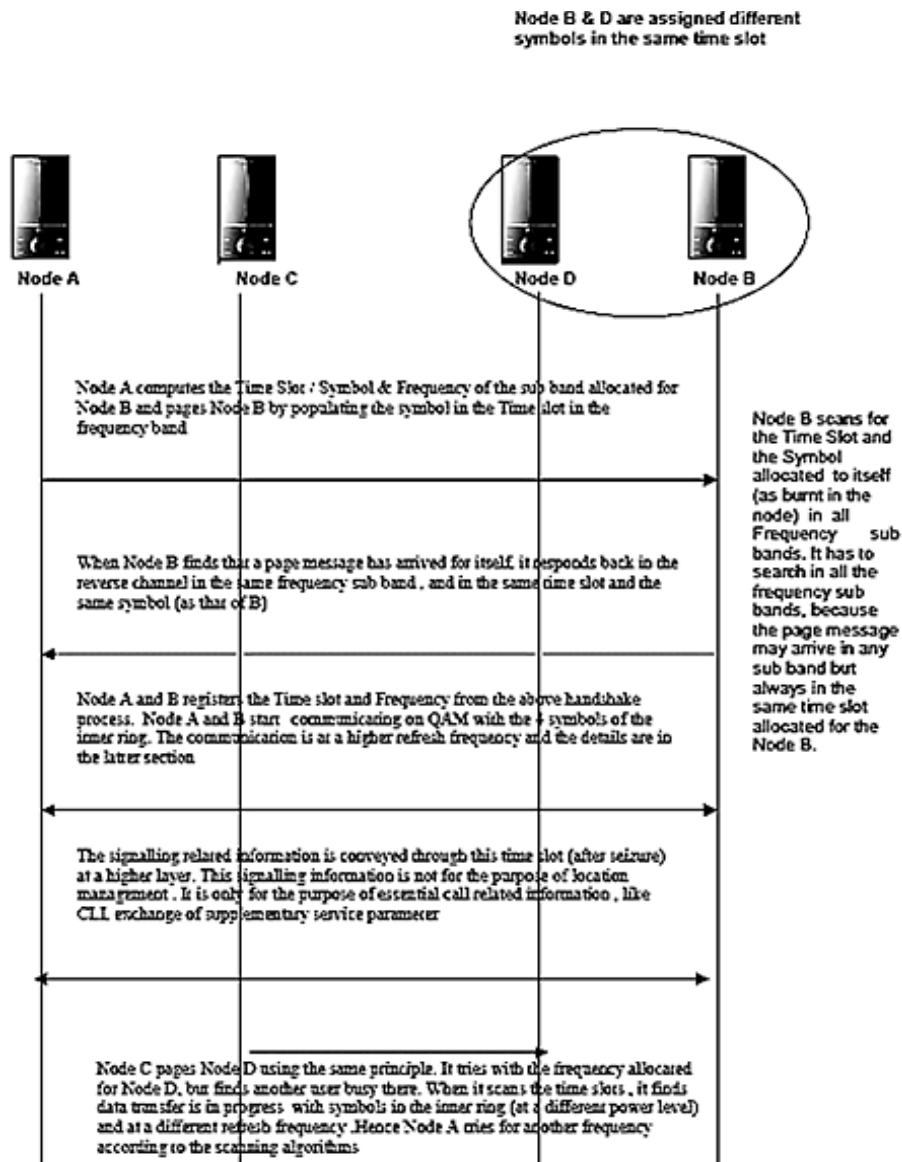


Figure 8: Handshake process between the mobile Nodes for Call Establishment

The above algorithm can be implemented in the Mobile Station as a firmware. The complexity of implementation of such algorithm in the mobile station and the penalization/impact on battery and processing power are discussed further.

Generation of time slot and symbol coordinate: The algorithm to generate the time slot and symbol coordinate from the dialed number during a call attempt is not intricate. A simple function can be defined which can be implemented as a subroutine in the main program. This algorithm should not consume significant processing and battery power.

Time Slot/Frequency scan/symbol coordinate: The time slot and frequency scanning is at present a part of mobile nodes. The scan algorithms are specific to modem device manufacturer. The additional

complexity is to scan for the specific symbol coordinate which acts like a page symbol. In the traditional approach this happens via layer 5/7 layer process which involves more processing resources.

2.12.1. Processes within the Mobile Station

The processes that are invoked within a mobile station for placing a call and then seizing a time slot for the traffic conveyance is detailed below in Fig. 9. The internal high level functional blocks of Mobile Station are also shown.

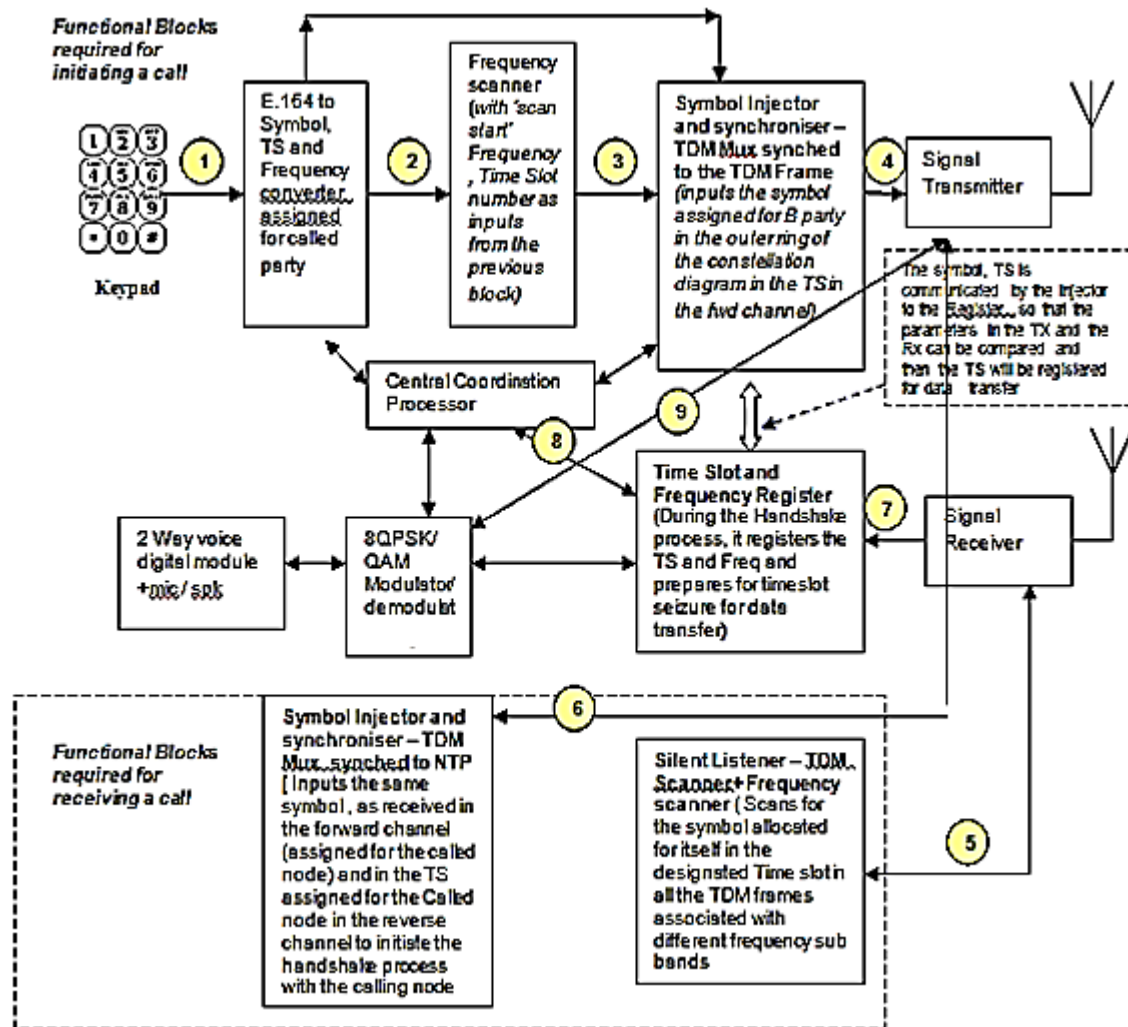


Figure 9: Internal Architecture of a Mobile Station pertaining to a Closed User Network

The descriptions of the processes as shown in the Fig. 9 are as follows. The processes 5 & 6 are related to the terminating leg of the call and will be invoked in the B Party Node. The rest of the processes (as in the diagram) are invoked in the Call Initiator, which is A Party.

Scenario (Mobile Node A calls Mobile Node B)

Processes In Mobile Node A (Calling Party)

Leg 1: The Called party number is input at the keypad. The numbers are fed to the section which analyses the Called number and results out the following parameters.

- Symbol
- Time Slot
- Frequency Band (from which the frequency scan will start to initiate handshake with Node B)

Leg 2: The three parameters are fed to the frequency scanner. The scanner sequentially scans the time slot in all the frequency sub bands available and determines which one is free. The time slot in the frequency band might be busy for data transfer amongst other Mobile nodes where the B party number has the same TS assigned but a different Phase (symbol). The frequency band information arriving from the previous block is used for commencing the frequency scan.

Leg 3: The TDM Mux synthesizes a TDM frame where the symbol is fed to the time slot.

Leg 4: The symbol in the specific time slot associated with the frequency band is transmitted on the air over the forward channel.

Processes In Mobile Node B

Leg 5: The Frequency / TS scanner scans the specific time slot allocated for itself, in all the frequency bands. It checks whether there is any symbol transmitted on that time slot in any bands. When it finds one, it interprets this as a page signal.

Leg 6: It formulates a similar TDM frame inputting the symbol in the same time slot (which is assigned for itself) and in the corresponding frequency band in the reverse channel.

Processes In Mobile Node A

Leg 7 & 8 & 9: The symbol is received and the TS and frequency band registered @ node A for data communication. It passes on this information to the coordination processor which interfaces with the QPSK modem to start full duplex voice communication using the registered time slot and the frequency band.

Processes In Mobile Node B

The above processes (7,8,9) are followed in mobile node B for seizing the Time slot in the frequency band for full duplex voice communication with B.

At a high level the following flowchart shows the Network Acquisition process between 2 mobile nodes in Fig 10.

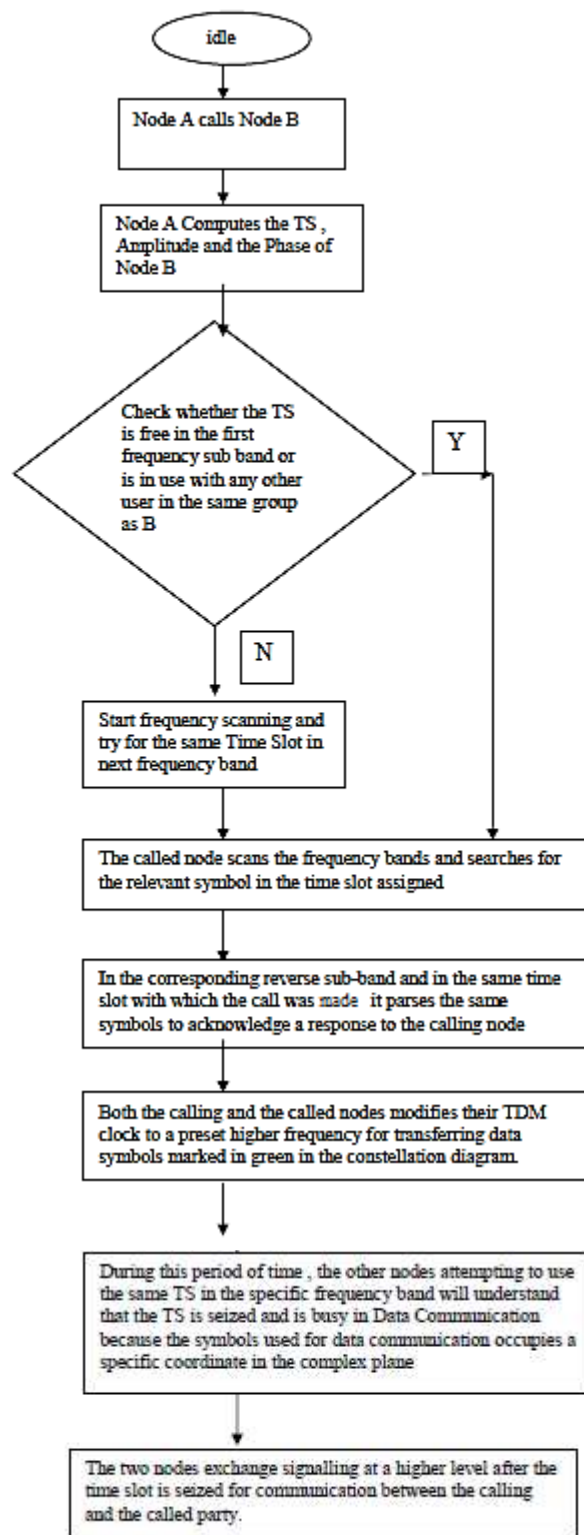


Figure 10: Algorithm for seizing a time slot for establishing traffic channel

2.13. Employing the physical layer modulation for addressing the nodes of a network – Simulation

The objective of this simulation is to demonstrate the feasibility of producing a set of symbols in the complex plane with fixed symbol coordinates representing multiple users. The model has been realized by placing 4 transmitters and 4 receivers separated by an AWGN channel. At the transmitting end we have 4 symbol generators, each one able to generate symbols pertaining to a specific coordinate of complex plane. At the receiving end, we position multiple filters, each one representing a mobile node. Each filter scans the incoming symbols and detects the ones pertaining to the specified coordinates. These symbols are related to addressing and not for conveying the data traffic. Hence the throughput is restricted and depends on the generic device paging frequency in the network. Paging will be done during any user operations, like call termination, message termination, network initiated data session setup, etc.

The rate at which these symbols will be transmitted will be minimal compared to the actual data transfer which is actuated by symbols transmitted in the inner ring. To exemplify, say 10 symbols per second for the 8PSK modulation (implying 100 Call attempts per second) corresponds to 36000 paging/ addressing attempts per hour per AFCRN. If 200 AFCRNs are derived from the allocated spectrum, this implies the support of 7200000 paging attempts in the network per hour.

As mentioned above, in this simulation, we focus only on the outer circle where the rate of symbol generation is not high. Hence BER calculation is not our focus for the time being. So we assume a generic AWGN channel in between the transmitter and receiver. But in latter section, we delve in radio network specific channel impairment scenarios when we discuss about coding in chapter 4. To summarise, the simulation is an attempt to establish that physical layer can be in fact directly linked to addressing (and hence indirectly linked to mobility management) which are otherwise higher layer functionalities in the State-of-the-Art network.

The inner ring is used for conveying the symbols for the given user, specific to the time slot. The symbol rate corresponds to the data speed required by the user. In the following simulation model, we attempt to portray this concept where the simulation represents only the outer ring meant for addressing. To keep the simulation simple, we have considered 4 users instead of 8 users. Each user is represented by 2 symbols for the purpose of redundancy.

In the model in Fig 14, we have 4 transmitters, and 4 receivers. Each transmitter transmits the symbols pertaining to specific coordinates on the complex plane. As there are 4 transmitters, hence the symbols have 4 coordinate sets, as can be seen in Fig 14, which represents the symbol output from the 4 transmitters. As said previously, we take 2 symbols per user for the purpose of redundancy, which can be seen the graphical output (in Fig 13.).

Each transmitter injects the symbols in specific Time Slots. Hence, we multiplex the time slots, carrying symbols of each users. This is demultiplexed at the other end by individual receivers. The receivers are designed to detect the information in specific time slot and specific symbol. For clarity, we will zoom the Transmitter and the Receiver part in Fig 11 and in Fig 12 respectively.

In Fig 11, we have a random integer generating the symbols at a periodicity. The random integer is fed to a data mapper, which maps only specific integers to fix the number of symbols per user. The TDM Multiplexer feeds the signal for multiple users over a Additive White Gaussian Noise (AWGN) Channel towards the receiver.

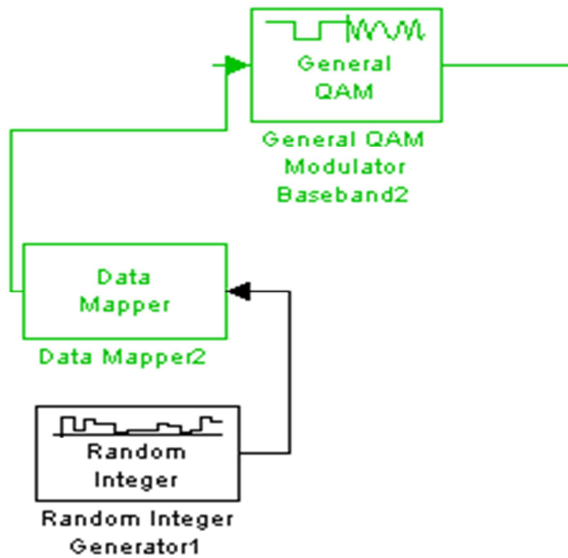


Figure 11: Transmission of the addressing symbols from a single user

According to the Figure 12, the receiver looks for specific symbols, for instance 3, 4 for the first transmitter module. Each receiver has a demultiplexer. The Demultiplexer at each receiver extracts the symbol/s meant for the specific time slot allocated for receiving the addressing information. The receiver node extracts the symbols only from the time slot allocated for the node. The 2 comparison blocks in Fig. 12 matches the symbols and if it finds the paging symbol allocated to the specific receiver, hence it outputs the symbol to the display.

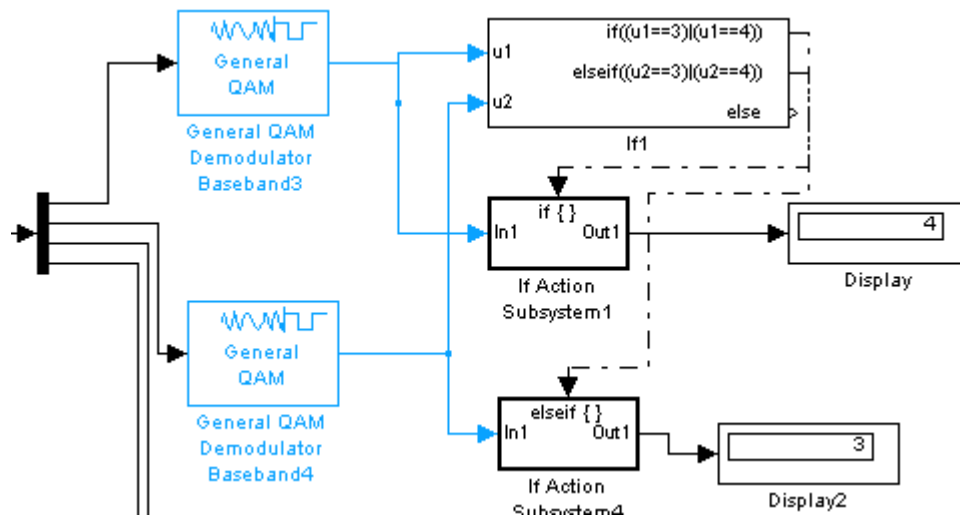


Figure 12: Receiver: Retrieval of the address symbols for a single user

We have 4 scatter plots (Figure 10) taken for the 4 users, where it is seen that the symbols specific to the user (2 per user in this case) is correctly received at the receiver end. In Figure 10, we are assuming an AWGN channel for the sake of simplicity. The effects due to the impairments in the physical channel at the radio network and the procedure of correction are defined in details in Chapter 4.

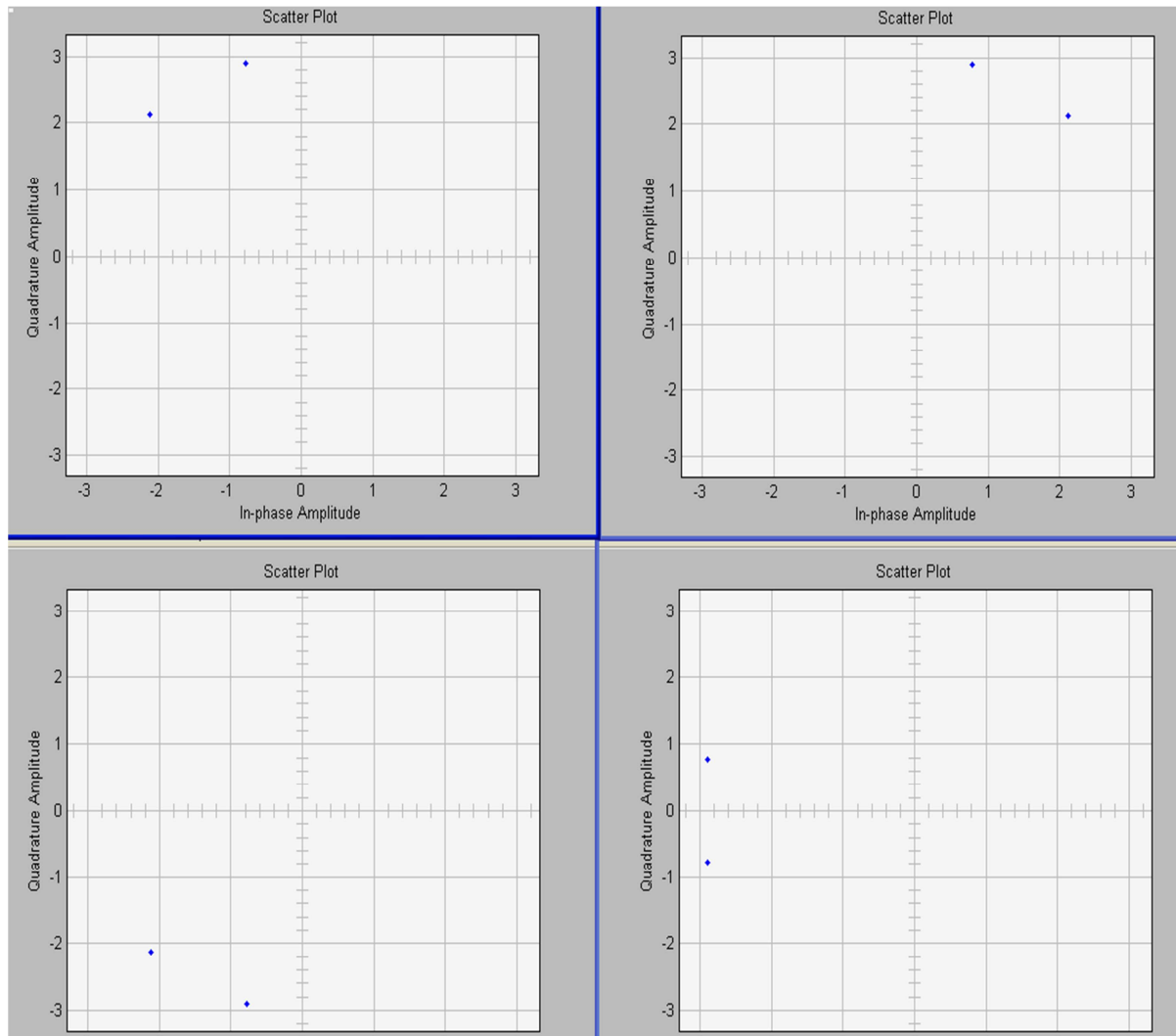


Figure 13:4 users in the complex plane

Figure 14 shows the complete model showing how the symbols are transmitted from the transmitter and then the symbols are multiplexed. The symbols for each user is displayed in each coordinate. The symbols are fed to the Demultiplexers at the 4 receivers. Each receiver has a demultiplexer to extract only the time slot allocated to the particular receiver and then feed the symbol to the next block for symbol detection.

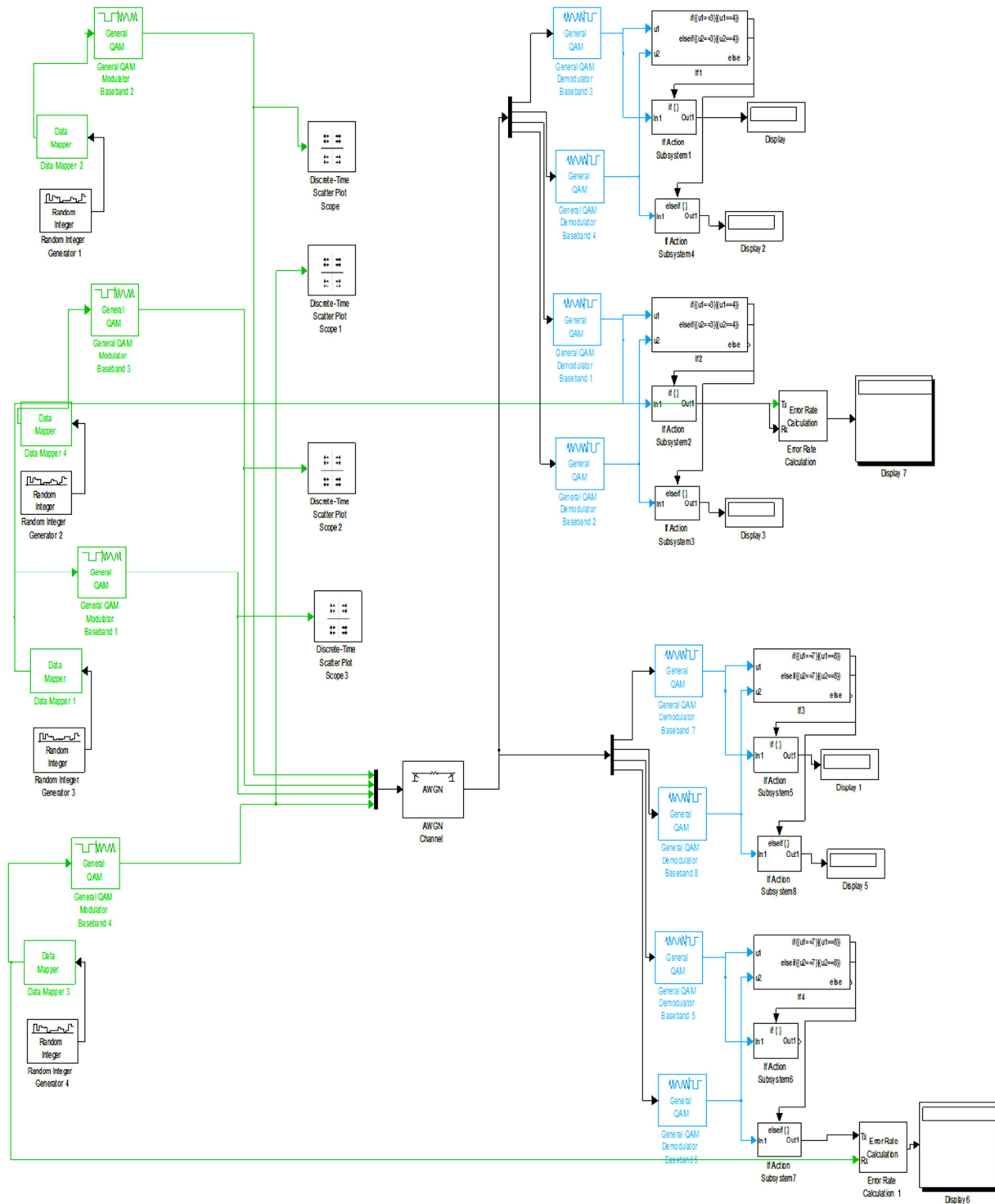


Figure 14: The simulation for 4 transmitters and 4 receivers system to portray the addressing mechanism using the physical layer addressing

2.14. Introduction of guard band

A guard band needs to be introduced between the AFCRN to ensure that the AFCRN transmit / receive simultaneously. In our proposed technology, a time slot pertaining to a specific AFCRN is first used for addressing and then for the data traffic.

The choice of the duration of the guard band solely depends on propagation delay caused due to increased distance between the mobile node and the access point.

For a typical GSM system, the slot lasts 577 μ s, which poses a limit on the range of 35 Kms.

If the range exceeds the threshold, the propagation delay becomes large compared to the slot duration and the mobile loses its time slot.

In order to avoid the situations, there are 2 way outs.

Introduce guard bands: This will increase the overall overhead.

Increase the tn, ie, the length of the slot: This will increase overall delay.

Now there are 2 types of time slots in consideration.

Tp => time slot for the main frame for addressing and identifying the user

Td => time slot for data

For the Tp, the system allocates a guard band only if 2 adjacent time slots carry one of the 8 PSK symbols of the outer ring meant for addressing and identifying the user. Or else the Guard band is provided if a Tp is used for voice channel (after addressing function is completed).

That is, if we see Td data bursts within Tp 1 and an adjacent Time slot Tp 2 (say), also carrying traffic information or the symbol for addressing, the guard bands will be placed between them (refer Fig 15)

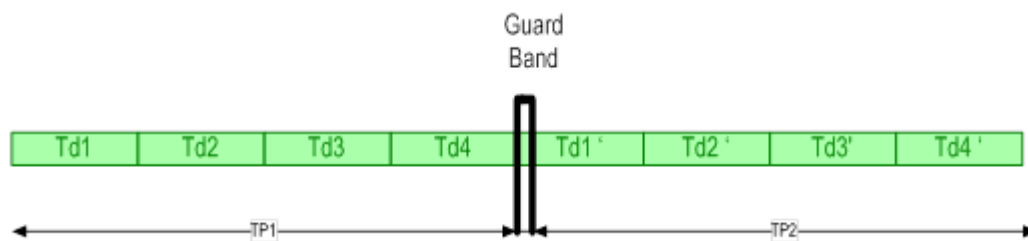


Figure 15: Guard band between two adjacent time slots used for data traffic

The Guard band is also placed between 2 frames to demark the different frames and avoid overlap. The guard band used to distinguish the frames is twice the length of the guard band used for the time slot Tp.

2.15. The Network Topologies in the light of the proposed Multiple Access Scheme

In this section, we will cover different network topologies and architectures where the proposed Smart Access Methodology finds its place. We start from a basic Closed User Network, then set out the architecture of a small sized network that we see in a town/ university campus/cruise ships, railways or Petroleum companies.

In our research, we have focussed primarily on three types of network topologies.

1. Closed User Network, with a fixed numbering plan.
2. Small sized network with open numbering scheme
3. A Global Roaming Network with open numbering scheme

This chapter covers the topic of Closed User Group Network and Small sized network. Next chapter covers the technicalities of the global roaming network.

2.15.1. Closed User Network

In a CUG, we have a private network where the addresses of the node will be known to all within the group. This implies that the mobile stations will have the functionality to interpret the parameters essential for channel acquisition and processing the call or other network operations. The main difference between a cellular network and the network realized by the proposed multiple access is that the mobility management related signalling operations are not prevalent. Hence there is no Location update process and the network is not aware of the presence or the location of the mobile station. For an outbound call (say), the channel acquisition for traffic information happens during the invocation of the call and not before that. On the receiving end, the mobile station listens for the appropriate signal at the right time slot which is known to itself during provisioning of the mobile station.

As discussed in section 2.2, the Mobile Node is identified in the network by an address following E.164 Numbering Plan. The basis of allocating an address for a mobile user however is dependent on the parameters that are directly related to the network implementation. For example, the principle varies from a Closed User Group network to a Global Mobile Network. The address allocation principle for a Closed User Group network is defined in section 2.2.

Depending on the size of the network, the number of addresses for the mobile node (according to subscriber population), number of channels and also the time slots per frame will change. So the length of the number (for the Mobile Node address) can be variable and so can be the three fields within that constitute the number (Symbol Number, Time Slot, AFCRN).

The various processes taking place internally in a mobile station starting from the point when the number is dialled and the channel is acquired for the call seizure is shown in Figure 16.

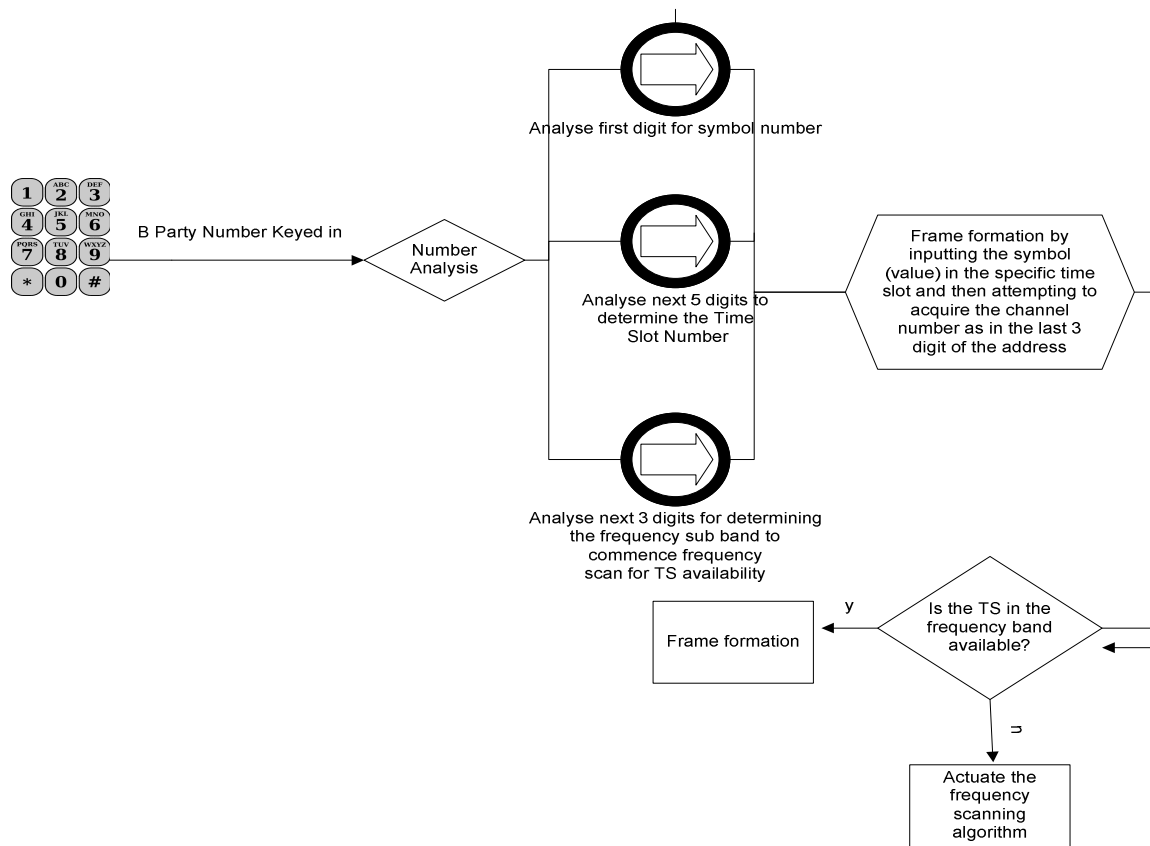


Figure 16: High level representation of frame formation after the destination number is dialed.

2.15.2. Small sized Mobile Network

For a mobile network, the functionality to analyse the Multiple Access related parameters by actuating Number analysis is present in the core network equipment (Service Control point) which is responsible also for AAA (Authentication , Authorisation and Accounting) . It does a validity of the profile based on the dialed number and, then analyses the parameters to be synthesized.

For a mobile network, there should not be any restriction for the number of subscribers that it can serve. The subscriber should be able to choose / change his number. Especially in the light of Number Portability, there should be no restriction to the address pattern, as the node address can actually belong to another network. Hence, the Identification number for a customer should be a E.212 address, instead of E.164. The E.164 address gets associated to the E.212 address in the HSS (Home Subscription System) or HLR (Home Location Register) in the core network, as is seen in the present mobile networks.

The high level components of the small sized mobile network are depicted in Figure 17. The RAN mainly comprise of the access points , which are repeaters which aggregate the forward and the reverse channel information to the Coordination processor , which is partly sitting on the RAN and partly at the CN. The Coordination processor is integrated with various core network elements, which are network adjuncts like HLR, SMSC and the Intelligent Network. So there is no major change in the core network elements and the signalling can still continue on the existing SS7 protocol, based on MAP and CAMEL / INAP etc. It

needs to be noted here that no signalling information is exchanged for the mobility management part. The signalling is mainly for AAA, tele-services, supplementary-services and value added services.

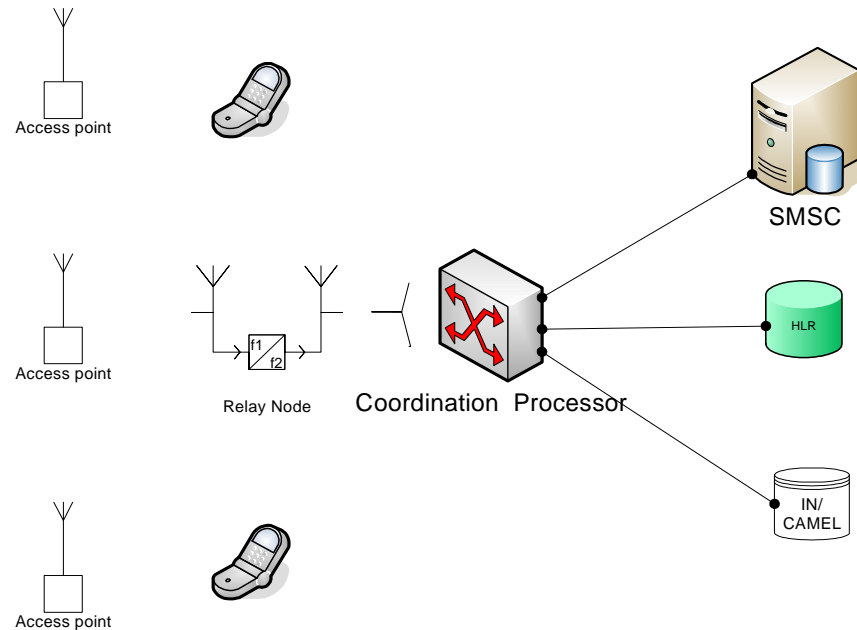


Figure 17: High Level Architecture of a small scale mobile network

The coordination processor is the heart of the entire network. It first aggregates the timeslots of a frame and constructs the frame in the reverse direction. After this, it analyses the time slots in the frame. It checks the coordinate of the time slot and then the symbol, the frequency and the frequency difference (last 2 digits) to derive the A party number for a particular time slot. At this point of time the B party number is not yet known. Hence it depends on the signaling level intelligence to obtain this information. Before it formulates and relays this symbol in the specific time slot (for the B Party) as it arrives, it checks in the AAA system to authenticate and authorize the A party customer.

For the frame formation in the downlink, the last 2 dynamic digits are not used, as no further digit analysis required at the network level.

The B party responds to the page message, after which the A party information and the other parameters arrive at the B party to initiate a ring signal.

2.15.3. Frame formation in the Downlink and Uplink

The Access points are also referred as Time Slot grabbers. In a location area, it receives the call attempts made by the Mobile stations in vicinity. This means, it looks for information arriving at specific timeslots. The information is mainly the symbol depicting a called user. From the different users the timeslot obtained are aggregated by a Mux before the coordination processors. If a timeslot is already busy in the pre call establishment activities, the coordination processor discards a second call request in the same timeslot towards the same or different user depending upon the symbol information.

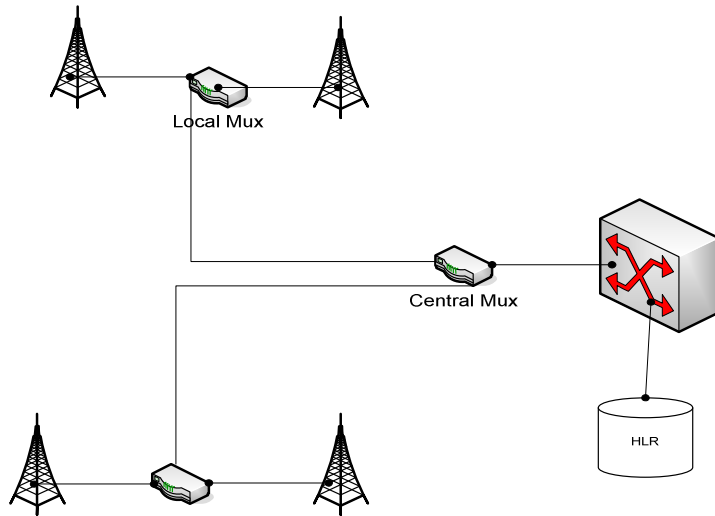


Figure 18: The Multiplexer Unit in the Access Network

As in figure 18, there can be several Multiplexers before the coordination processor. But just before the coordination processor, there is an intelligent central multiplexer which aggregates all the time slots, and formulates the frames to be further processed by the coordination processor.

2.15.3.1. Frame formation in the Uplink

The control signalling messages are invoked by the mobile device only when it initiates a network operation, like Mobile origination Calls, SMS etc. But no messages are generated for Location Update unlike the State-of-the-Art. The device never gets attached to the network. The network does not keep a track of the location and presence of the device continuously, hence we eliminate the need of the continuous exchange of mobility management messages between the network and the device.

Fig19. Illustrates how the frame is formed in the uplink for the addressing part. Multiple User Equipments (mobile devices) as in this diagram invoke call requests. The UE (User Equipment) first hunts for an AFCRN where the time slot in the allocated position (as assigned to the user during the provisioning process) is identified. The symbol is then populated in the time slot and the UE generates a time frame by injecting only the specific symbol in that time slot in the time frame. Other time slots in the time frame are empty (as in Fig.19). The time frame pertaining to the available AFCRN is marked in black in Fig.19 which implies that the User Equipment has successfully populated the assigned symbol (assigned to itself) in the time slot predefined for the User. The Access point transfers the information to the local Multiplexer which aggregates the symbols ingressing from the User Equipments and formulates a single time frame for each AFCRN. The Local Multiplexer formulates a time frame by injecting the symbols from the different users in the appropriate slots as per the time frames arriving from the User Equipment. This time frame is transported to the Aggregate Multiplexer. The Aggregate Multiplexer synthesizes an aggregate time frame w.r.t each AFCRN and forwards it to the Coordination Processor for analysis and call processing.

At this stage, it is impossible to identify the address of A party (calling number of a particular user) from the symbol coordinate and timeslot, because the AFCRN is variable. Hence the Coordination processor cannot commence authentication. It first needs to secure the Time slot in both Uplink/Downlink

directions between the MS and itself. Hence it generates a response message in downlink towards the MS using the same AFCRN number that was used by the A party in the during the uplink layer 1 message for call initiation. However, the particular AFCRN may not be available because it may already be in use as traffic channel by other users. In this case, the Coordination Processor does not generate the response message towards the Mobile Station. The MS waits for the timer expiry (in milliseconds) and restarts the scan process and re-initiates a new message with another available AFCRN. In this process, it skips the AFCRN that was previously determined as unavailable.

Following a successful layer 1 handshake between UE and the Coordination Processor, the Time Slot pertaining to the AFCRN is used further as traffic channel. The UE generates a layer 7 message towards the Coordination Processor encapsulated in the time slot that was allocated following the layer 1 handshake process as described earlier. The data conveyance materialises following M2 PSK modulation based on the inner ring of the constellation diagram. UE populates A party addresses (calling address in E.212 format), authentication parameters, dialled numbers, supplementary service information in this message, so that the Coordination Processor can liaise with the HSS/HLR to actuate authentication, authorisation for A party. Subsequently, it analyses the B Party (dialled) number and figures out the outgoing channel by carrying out dialled digit analysis.

The above process deviates from mobile originating call procedure followed by the State-of-the-Art Mobile Networks. Generally in conventional networks, the Serving MSC/MME/CSCF [10][23][24] generates a ISUP/SIP message directly for call termination towards the B party address. There is no interrogation towards the HLR/HSS. Only in case of Home routing scenarios, the call is routed towards the Home network where the SCP (Service Control point) is queried to generate a 'Connect To' number.

2.15.3.2. Frame formation in the Downlink

Fig 20 describes how the call matures at the downlink towards B Party (called address) UE. When the call arrives at the Coordination Processor, it firstly analyses if the B party address is served by the same mobile network, ie, the smart network. If yes, then it interrogates the HSS/HLR for obtaining the symbol coordinate, Time Slot and the Primary AFCRN allocated for the B Party. The supplementary and the tele-services assigned for the B party are also downloaded by the HSS/HLR towards the Coordination Processor.

Subsequently, the Coordination Processor injects the symbol of B party in Time slot (preconfigured for B Number) in the time frame for the particular AFCRN in the downlink. In case the primary AFCRN is busy, then alternative AFCRN's are tried according to the predefined frequency scanning schemes. Note that this time frame also carries the symbols from other users in the respective time slots. Aggregate frame is formed and is routed to the aggregate multiplexer. The aggregate multiplexer broadcasts the time frames pertaining to all the AFCRN's in the Downlink via the Local Multiplexer and Access Points. B party scans the AFCRN's and looks out for the specific symbol contained in the specific time slot assigned for B number. If it finds one, then it formulates a response message by injecting the same symbol in the same time slot position in the UL. If the time slot (pertaining to the AFCRN) is busy, there is no response back from B party. The Coordination processor has a time out and tries to page B party again with an alternate AFCRN according to the frequency scanning logic. As AFCRN is not fixed we cannot ascertain at this stage seeing the radio layer physical layer message whether the page message is intended for the particular B party address or some other users. The page message can indeed be meant for another user provisioned with the same symbol coordinate and time slot position. Hence, multiple users can simultaneously try to generate the response message. But once the coordination processor receives the layer 1 page response, the time slot related to the AFCRN will be reserved for use by the coordination

processor. The time slot / AFCRN is also reserved at the UE end.. The subsequent layer 7 message generated by the Coordination processor will bear the B party address (E.212) as well as the A Party address (E.164 for the CLI) and the supplementary service information. The specific UE (the original B party) will respond to this layer 7 message. The other UEs which reserved the Time slot will release it when it ascertains from the layer 7 message carried by the Time Slot that the page was intended for another user (ie the B party).This completes the addressing process in the downlink.

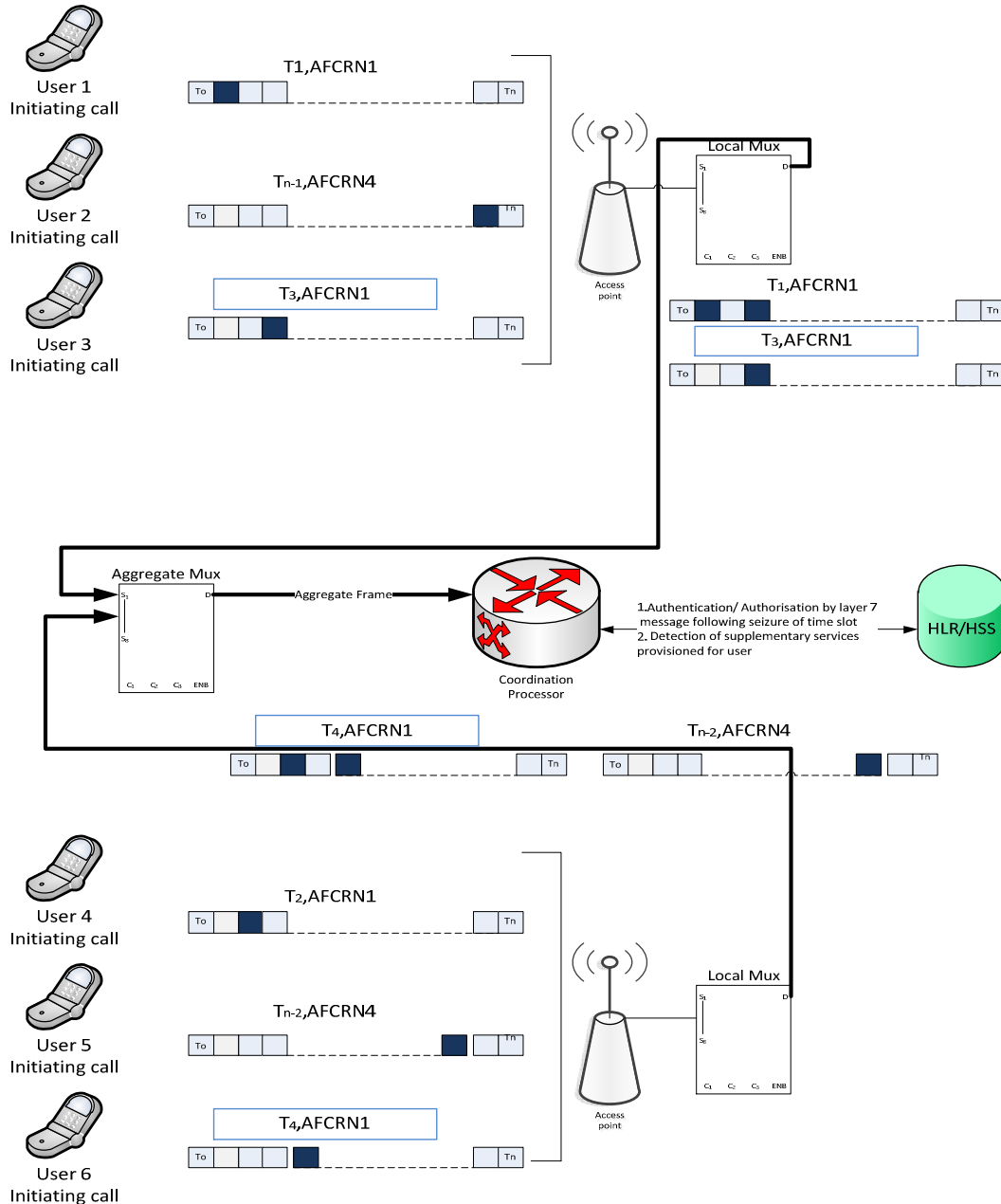


Figure 19: Aggregate Frame Formation in the Uplink and user (A Number) identification by Coordination Processor

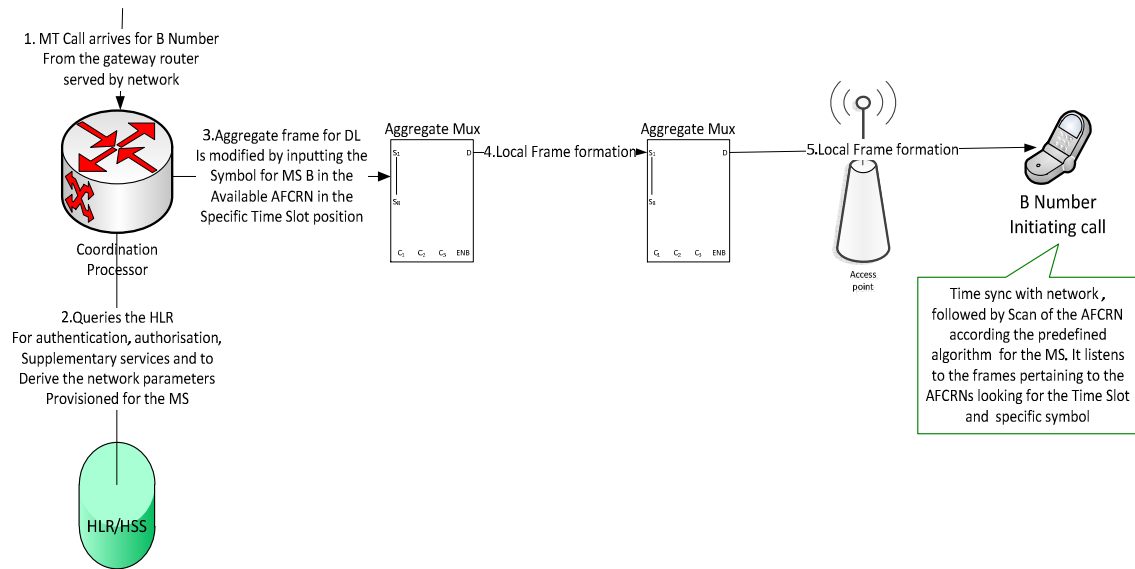


Figure 20: Frame Formation in the downlink

2.16. The synchronisation mechanism

Synchronisation of the time frame between the Mobile Station, Access point and the Coordination Processor is an important factor.

The timing synchronisation pulse will be generated by the coordination processor which in turn is synchronised with the atomic clock. The AP (Access Point) sends signals to enable the MS to synchronize itself to the AP and if necessary correct its frequency standard to be in line with that of the AP. The signals sent by the BTS for these purposes are:

- a) Frequency correction bursts;
- b) Synchronization bursts.

The timings of timeslots, TDMA frames, are all related to a common set of counters which run continuously whether the MS and AP are transmitting or not. Thus, once the MS has determined the correct setting of these counters, all its processes are synchronized to the current serving AP. The MS times its transmissions to the AP in line with those received from the AP. The BTS sends to each MS a "timing advance" parameter (TA) according to the perceived round trip propagation delay AP-MS-AP. The MS advances its timing by this amount, with the result that signals from different MS's arriving at the BTS and compensated for propagation delay. This process is called "adaptive frame alignment".

Access Point (AP): Requirements for Synchronization

The AP shall use a single frequency source of absolute accuracy better than 0.05 ppm for both RF frequency generation and clocking the timebase. The same source shall be used for all carriers of the AP. APs shall be synchronized together such that the timing difference between different APs shall be less than 1 symbol period.

Internal AP carrier timing

The channels of different carriers transmitted by a BTS shall be synchronized together, i.e. controlled by the same set of counters. The timing difference between the different carriers shall be less than 1/4 symbol periods, measured at the AP antenna.

Initial Timing advance estimation

When the AP detects an access burst transmission from a MS, it shall measure the delay of this signal relative to the expected signal from an MS at zero distance under static channel conditions. This delay, called the timing advance, shall be rounded to the nearest symbol period and included in a response from The BTS when applicable.

Maximum timing advance value

The maximum timing advance value TA max shall be 63. The TA value is normally between 0 and 63, with each step representing an advance of one bit period (approximately 3.69 microseconds). With radio waves traveling at about 300,000,000 meters per second (that is 300 meters per microsecond), one TA step then represents a change in round-trip distance (twice the propagation range) of about 1,100 meters. This means that the TA value changes for each 550-meter change in the range between a mobile and the base station. This limit of 63×550 meters is the maximum 35 kilometers that a device can be from a base station and is the upper bound on cell placement distance.

Delay tracking

The AP shall thereafter continuously monitor the delay of the normal bursts sent by from the MS. If the delay changes by more than one symbol period, the timing advance shall be advanced or retarded 1 and the new value signalled to the MS.

Synchronisation initiated by the MS:

MS carrier frequency

The MS carrier frequency shall be accurate to within 0.1 ppm, or accurate to within 0.1 ppm compared to signals received from the AP. These signals will have an apparent frequency error due to BTS frequency error and Doppler shift. The MS shall use the same frequency source for both RF frequency generation and clocking the timebase.

Internal timebase

The MS shall keep its internal timebase in line with that of signals received from the AP. If the MS determines that the timing difference exceeds a set time duration in seconds, it shall adjust its timebase in steps .

Timing of transmission

The MS shall time its transmissions to the BTS according to signals received from the AP. The MS transmissions to the BTS, measured at the MS antenna, shall be 3 timeslots-TA behind the transmissions received from the BTS, where TA is the last timing advance received from the current serving BTS. The tolerance on these timings shall be ± 1 symbol period.

Synchronisation initiated by Coordination Processor

The Coordination Processor is synchronised with the Aggregate Mux / Local Mux which are interfaced with the Access Points distributed across the network area. The Network Time Protocol (NTP) clock is used to synchronise the RAN elements connected to the coordination processor. The frame formed by the local Multiplexer and then aggregated by the aggregate Mux is fed to the coordination Processor (and vice versa) . The Coordination processor and the MUXes have the Frame synchroniser component which is dedicated for Frame synchronisation.

A dedicated timeslot is assigned for frame synchronisation, as represented by FS in Fig 21. This slot is followed by the Subframe ID (SFID) which is subsequently followed by data packets. The frame synchronizer recognizes this pattern and aligns the data into Minor Frames or Sub Frames [ref Fig 21].

FS	SFID1	D1	D2	D3	D4	D5	D6	D7	D8
FS	SFID2	D1	D2	D3	D4	D5	D6	D7	D8
FS	SFID3	D1	D2	D3	D4	D5	D6	D7	D8
FS	SFID4	D1	D2	D3	D4	D5	D6	D7	D8
FS	SFID5	D1	D2	D3	D4	D5	D6	D7	D8

Figure 21: Frame Synchronisation

2.17. The functionality and the architecture of the coordination processor

The Coordination processor [1][13] replaces the conventional time/space switch of a mobile network. So it interfaces with the Aggregate Mux and access points to convey both the information in the forward and reverse channels. On the other hand, it interacts with the core network elements.

As discussed in the above sections, we have an understanding how the Coordination Processor takes part in frame formation related functionalities. In this section, we will be focussing more on the internal processes and functionalities of the coordination processor. The coordination processor sits between the access and the core network. It replaces the functionality of a modern day space and time switch of a mobile network.

The various functionalities of a coordination processor are listed below.

1. Deriving the addressing related intelligence of the destination node from the reverse channel.
2. Populating the information (symbol) in the time slots and frame formation in the forward channel.
3. Interaction with the core network for AAA.
4. Processing the signalling information arriving at the forward channel and formulation of the signalling information in the reverse channel.
5. Functionalities related to the supplementary services for the users.
6. Billing / CDR related processes.
7. Various other OSS/BSS related functionalities.
8. SS7 signalling and PCM Trunks with external legacy network.

The primary functionality of the coordination processor as explained above, and its interworking with the different functional components are shown in Fig 22.

The high level architecture of the coordination processor is shown below, which is self-explanatory:

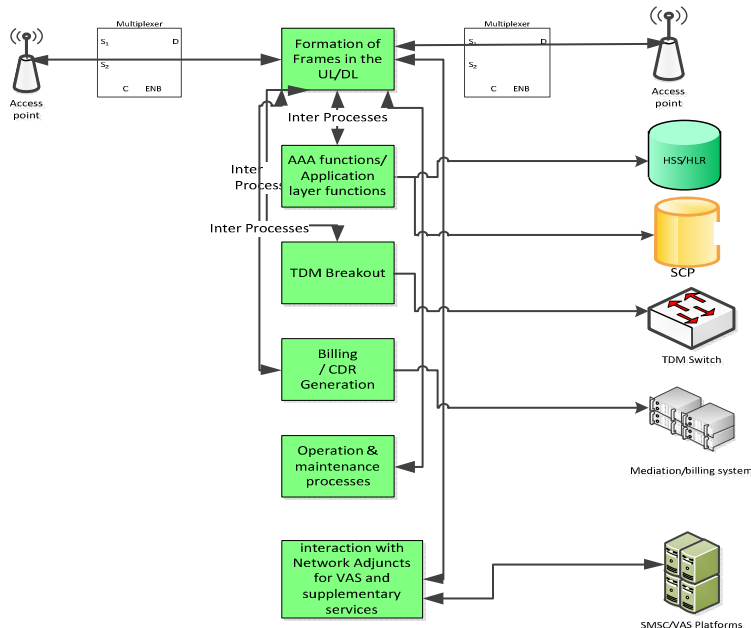


Figure 22: The Coordination Processor

The functional components in the coordination processor related to frame formation in the forward and reverse channels are detailed in the process flow as per Fig 23.

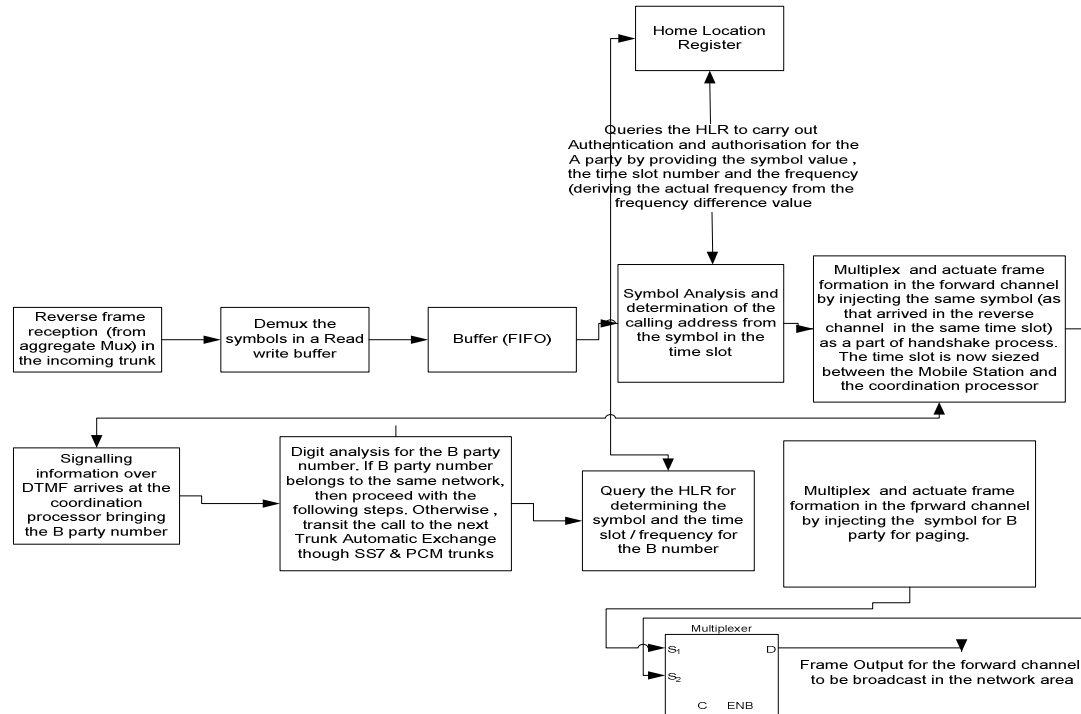


Figure 23: Frame formation aided by Coordination Processor

The functional components in the coordination processor related to dialled number analysis and determination of the output route is provided in Fig 24. The Coordination processor identifies the type of call based on the dialled digits and accordingly selects the appropriate route to transit the message. The coordination processor routes a call after digit translation through the appropriate route that is defined for the dialled number.

Fig 24 identifies the different call types processed by the coordination processor. All these processes require digit analysis and translation. After formulating the desired digit set (in case of group calls or call conferencing) the Coordination processor actuates the formation of the time frame following digit collection for all the bi directional call attempts.

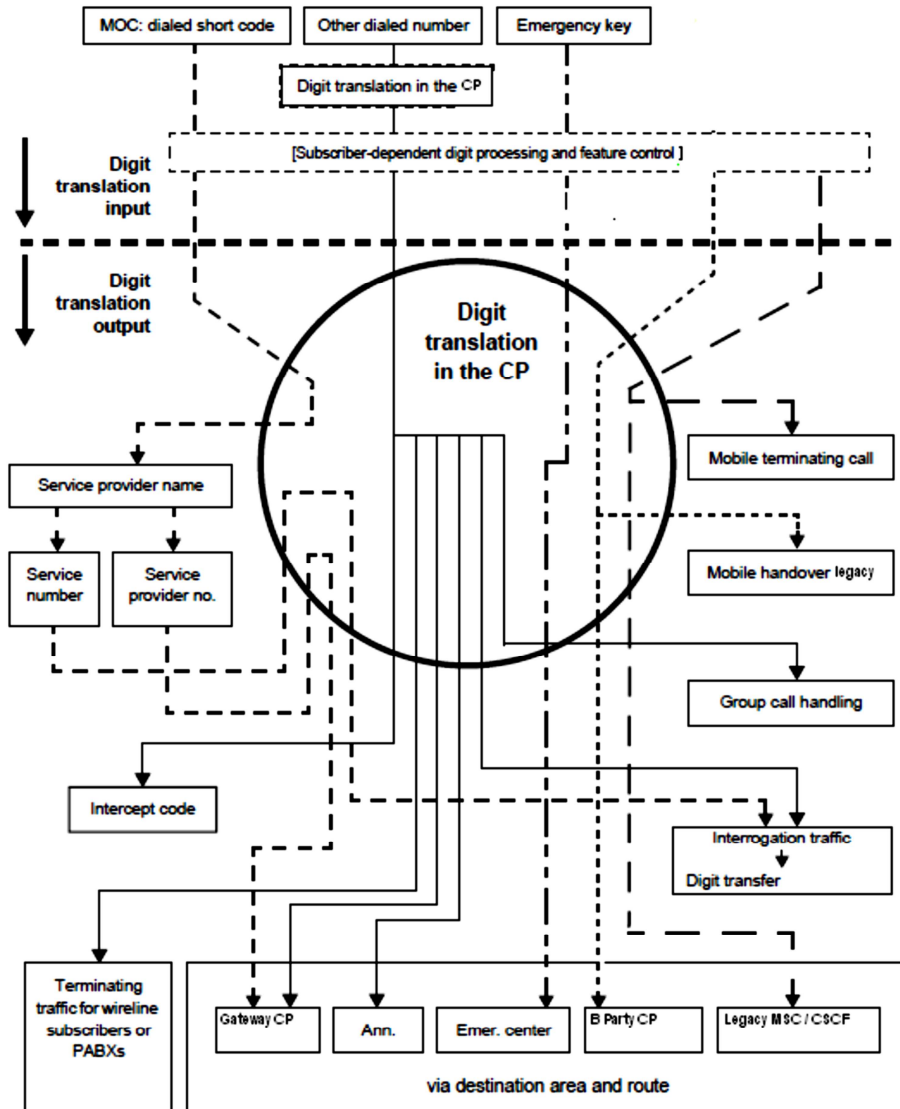


Figure 24: Call Handling in Coordination processor

Processing of the TDMA frames by Coordination Processor

As defined previously in section 2.2, Fig 3, the TDMA frame comprise of 2 main time slots.

1. T_p is time slot which represents an user in the frame
2. T_d is the time slots pertaining to data samples carrying the user traffic.

The t_p time slots arrive in bursts. The arrival gap depends on the size of the time frame. At the access point, the TDMA frames on all of the radio frequency channels in the downlink shall be aligned. The same shall apply to the uplink. At the access point, the start of a TDMA burst for traffic channel (represented by T_d) on the downlink is delayed by the fixed period of 3 timeslots from the start of the TDMA frame on the uplink. This is shown in Figure 25. The data traffic part is handled by the QPSK

of the inner ring of the constellation diagram. The time slot duration for the data traffic is 8 times less than the one which is involved for node addressing.

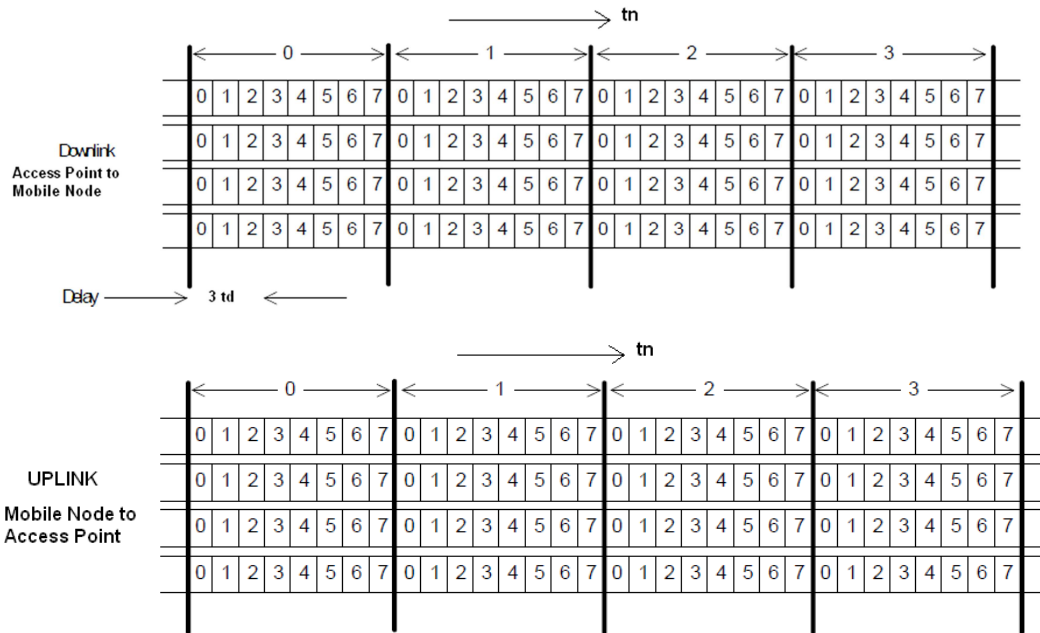


Figure 25: Downlink with a delay of $3td$

At the mobile station this delay will be variable to allow adjustment for signal propagation delay. The process of adjusting this advance is known as adaptive frame alignment. The staggering of TDMA frames used in the downlink and uplink is in order to allow the same timeslot number to be used in the downlink and uplink whilst avoiding the requirement for the mobile station to transmit and receive simultaneously. The period includes time for adaptive frame alignment, transceiver tuning and receive/transmit switching.

The timings of timeslots, TDMA frames, the rotation of time groups are all related to a common set of counters which run continuously whether the MS and Access Point (AP) are transmitting or not. Thus, once the MS has determined the correct setting of these counters, all its processes are synchronized to the AP.

The MS times its transmissions to the AP in line with those received from the AP. The AP sends to each MS a "timing advance" parameter (TA) according to the perceived round trip propagation delay AP-MS-AP. The MS advances its timing by this amount, with the result that signals from different MS's arriving at the AP and compensated for propagation delay. This process is called "adaptive frame alignment".

2.17.1. End2End Call flow

Fig. 26 which is self-explanatory demonstrates the transactions and the call trajectory between the mobile devices.

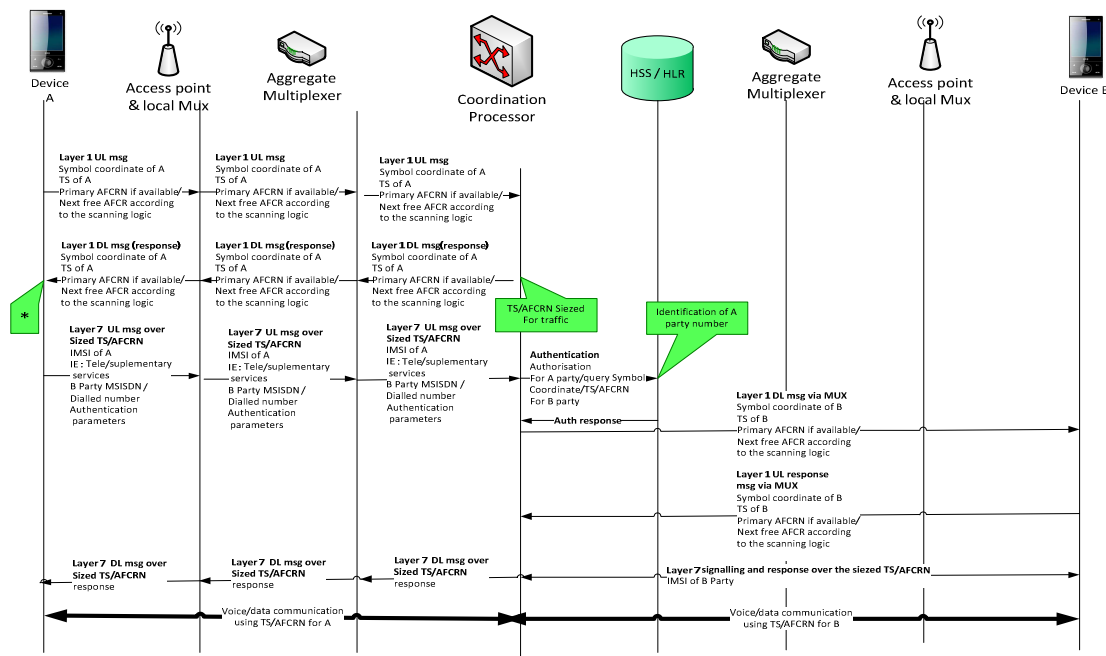


Figure 26: End2End call flow between 2 mobile devices actuated by the Coordination Processor

2.17.2. Capacity calculation for a small sized network not actuating frequency reuse:

The number of channels must be related to the number of carriers (frequencies) available:

- 8 channels (timeslots) per carrier
- The number of AFCRNs = 124.

In particular the parameters that the applicable in the Location Area are:

- Mean call holding time = 90 seconds
- Grade of Service = 1 %
- Total number of available carriers = 124

The number of subscribers that the Network can support is :

- Mean call holding time of 90 s implies the average traffic per subscriber is 25 mE
- Using Erlang B tables for GoS = 0.01 and $n = 124 \times 8 = 992$ channels, gives 666.589 E = 666589 mE.
- Number of subscribers that can be supported is: $666589 / 25 = 26663.56$.
- The maximum range of a cell is 35 Kms. This is because the Timing Advance parameter is set to 63.

We keep the TA parameter used in GSM to make sure that the width of the time slot stays at $577\text{ }\mu\text{s}$. This is done to make sure that the same error codes (convolution/fire/block) which are used for GSM, that are tried and tested over the years are reused in the present proposal.

Number of Access points helps in boosting the signal strength (acting as repeater) but does not participate in forming a cell. Hence this factor does not affect the spectral efficiency. But it affects the distance between the antennas of the access point and the mobile station. One of the shortcomings of the proposed mechanism is the network scalability. These types of networks can be functional for small MVNOs, University campuses/ factories/ cruise ship etc. In the next chapter we set out architecture with smart location management scheme for global roaming network.

4

Deploying the Novel Multiple Access Scheme for a Global Mobile Roaming Network

3.0. Introduction

The state-of-the-art mobile network has rooms for improvisations especially in the domain of mobility and addressing. With all the evolving technologies, we primarily see two trends [10][23][24].

- Evolving towards an all IP architecture
- Realise higher bandwidth and latency to cater the next generation mobile phone applications.

But mobility and addressing in the state-of-the-art has virtually no deviation from the legacy schemes.

In this chapter we discuss the processes to implement our novel multiple access topology for a global roaming network and introduce unique mobility management scheme.

First we focus on the proposed mobile network architecture and discuss its alignment and compatibility with the proposed multiple access scheme. The higher layer processes encompassing mobility and location management and addressing are discussed. We discuss the novel process of mobility management which is driven more by the mobile station. Further, we discuss the checker board approach based cell design and elaborate the handover process. In Fig 27, we take the example of Belgium, where we attempt to allocate square shaped cells according. Note that this depiction is only for the purpose of exemplification. The allocation of the cells in the Location Area (ie. Belgium) may not be the actual implementation that will be followed for deployment. It is just an example.

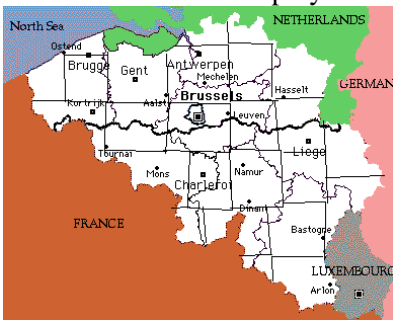


Figure 27: Network area of Belgium segmentised in large sized square shaped cells

In this chapter, we compare with the present day mobile technologies and also highlight the primary advantages of the proposed novel mobile roaming network over the prevailing technologies.

3.1. High Level description

For a global sized network, reusing the frequency is mandatory for a higher spectral efficiency. Frequency reuse also implies that we have considered an important process related to mobility management called handover.

In a present GSM / UMTS network, substantial bandwidth resource is consumed due to signalling. Also the complexity in the application level processes need to be considered, as MM and RRM are layer 7 activities. This is even true for the state-of-the-art 4G (LTE) networks, where eNodeB [24] in conjunction with MME is responsible for MM.

We thrive to handle frequency allocation and reuse through a process which is more driven by the intelligence in the handset and the physical layer.

3.2. Mobility and Handover

The geographical area is distributed in multiple cells. The cells are square shaped each having a diagonal distance from the centre of around 35 Kilometers. The 35 Kms restriction is due to retaining the timing advance value of max 63. As timing advance changes every 550 metres, hence the max cell size is $550 \times 63 = 35$ Kms.

There is a central access point with a coordination processor in each cell.

Or there may be a central system with virtual coordination processors taking care of each cell.

Figure 27, demonstrates the architecture and cell allocation pattern for the access network [1]. It also describes the handover functionality to be incorporated in the proposed network and focuses on the control logic in the Mobile Station for initiating handover.

The entire network area can be represented as chess board. Each block represents a square shaped cell. The cells do not have any cell ID. So the network is agnostic to the exact location of the mobile station. The white blocks are allocated AFCRN from 1 to 64, while the black ones are allocated AFCRN from 65 to 124.

Each mobile station is fitted with two transceivers. There is an active part, which is active in cell which serves the mobile station. The dormant part keeps on scanning the adjacent cells and is mainly responsible for actuating the handover procedure. It only scans the available timeslots (in its domain) in the subsequent frequency range. Hence while the active one operates in the AFCRN domain 1 to 64, the dormant transceiver operates in the range 65-124.

A central processor in the Mobile Station monitors and compares the signal strength as perceived by the 2 transceivers. The MS also has a inbuilt GPS unit, and if in open air (with LOS with the satellite) can also determine the direction and speed of motion. It also has an electronic compass to analyse the direction and interpret the direction of motion if the LOS with the satellite is not available. Analysing the signal strength of the 2 received signal (and also gathering intelligence from the GPS / compass unit if available), the handset decides to initiate the handover,

In such a case, the passive Tr/Rcv informs the time slot and the frequency which is in use by the active Tr/Rcv. The Coordination Processor (CP) serving the cell establishes a channel (TS) , the same value which was intimated by the passive Tr/Rcv. This happens exactly when the active Tr/Rcv releases the channel so that the CPs of the adjacent white

and the black boxes can establish channel on the same TS / Freq which is currently in use between the CP of the white Box and the MS.

Now the Physical path of the call is the Passive Tr/Rcv of the MS \leftrightarrow to \leftrightarrow CP/Access point of Black cell to CP /Access point of the white cell which was already in conversation phase directly with the MS before the handover.

After the handover, the passive Tr/Rcv in the MS becomes the primary one, and the Tr/Rcv which was primary before becomes dormant and starts scanning the pilot strengths of adjacent cells to check the possibilities of further handovers. The handover never happens between the cells with the same AFCRN band that is with the same colour. The lateral handover is performed directly between cells of opposite shades. However, the diagonal handover also happens through a process of 2 lateral handovers. Fig. 28 demonstrates the two modes of handovers.

The algorithm to be followed by the Mobile Station to actuate and control the handover process is in Fig 29.

The process of handover is described in details in the following diagrams.

- Block Diagram of the Mobile station related to handover initiation
- Algorithm for the handover
- Call Flow for Handover.

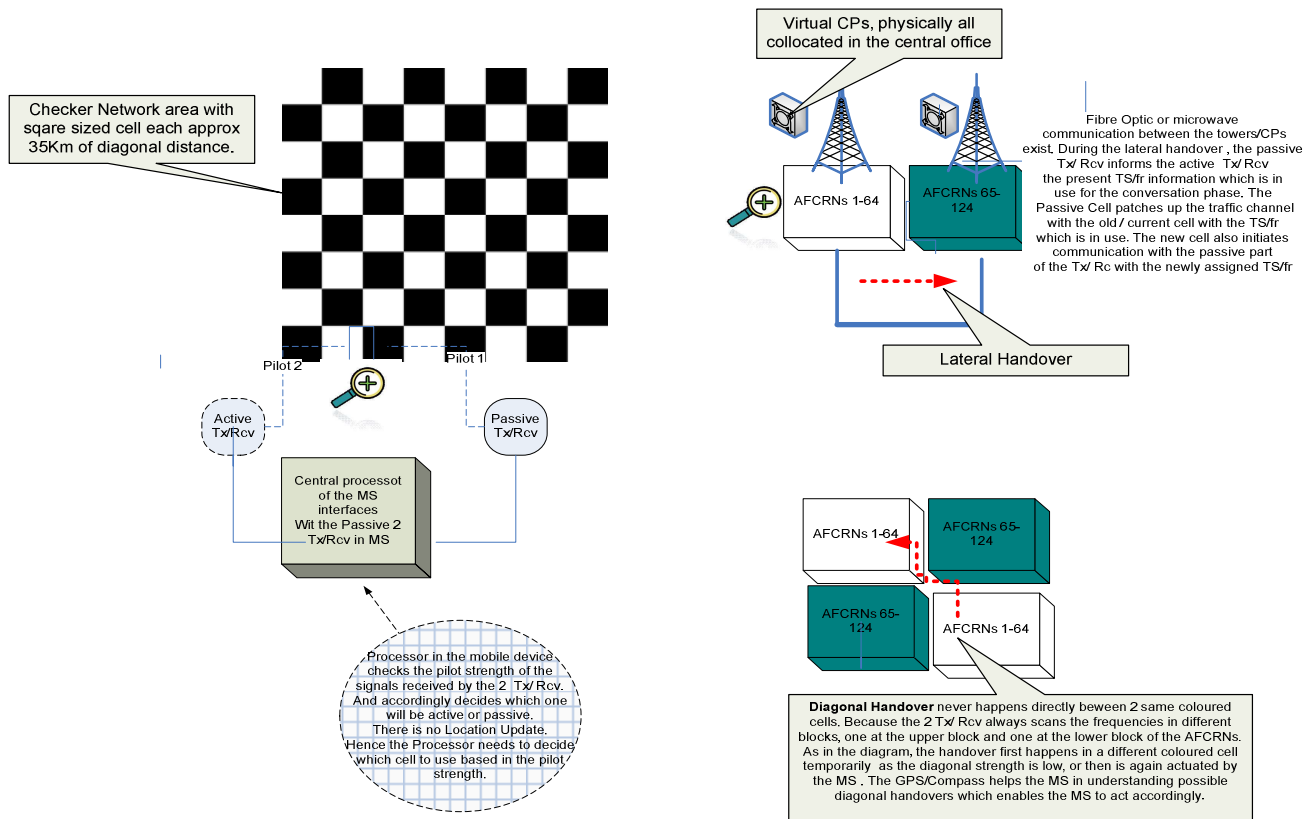


Figure 28: Handover Initiation by the Mobile Station

3.2.1. Algorithm for handover

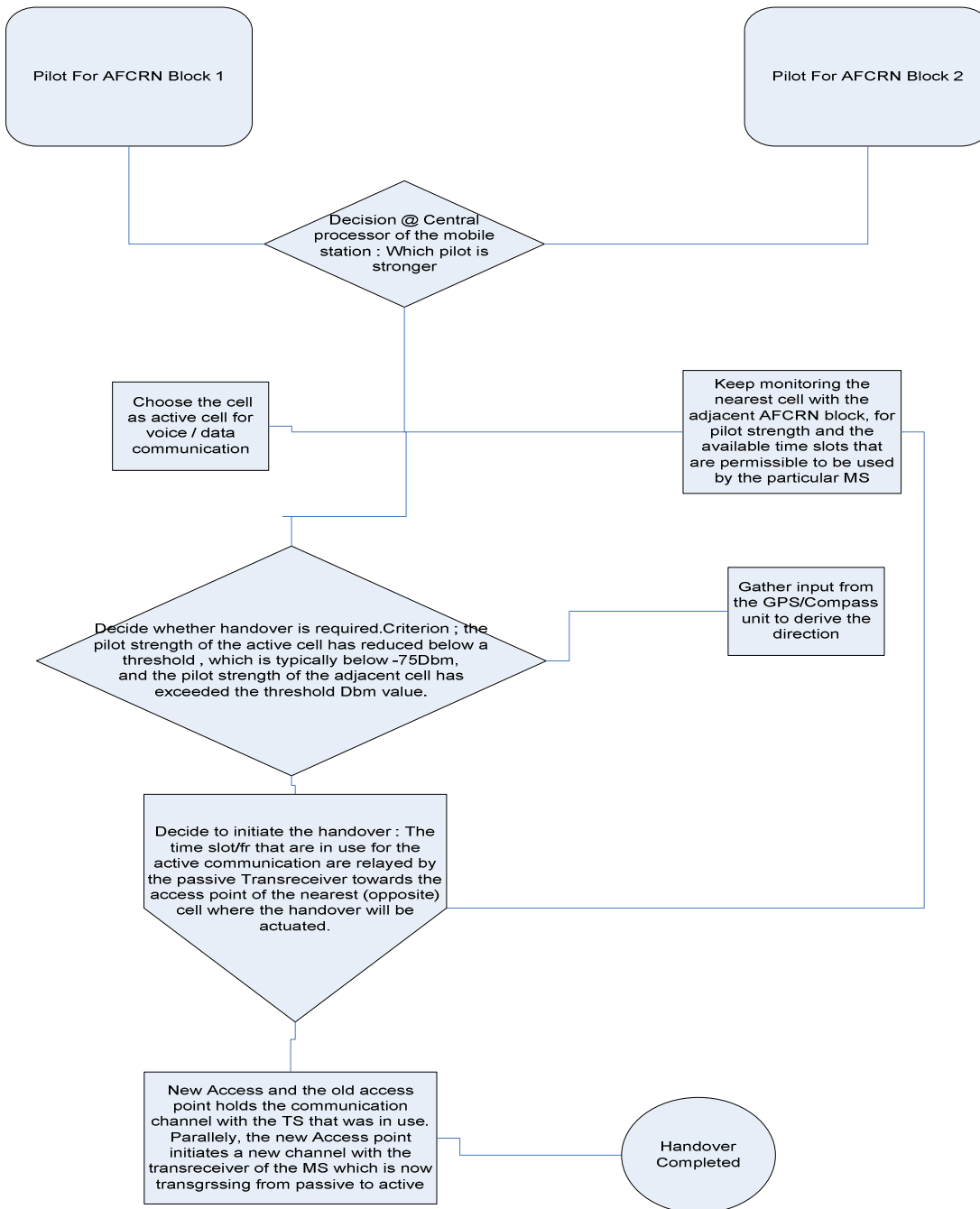


Figure 29: Algorithm for handover

3.2.2.Call Flow for the cell handover

The complete call flow for the Handover process initiated by the Mobile Station and its interaction and handshake process with the Coordination processor / Access Point is detailed in Fig 30.

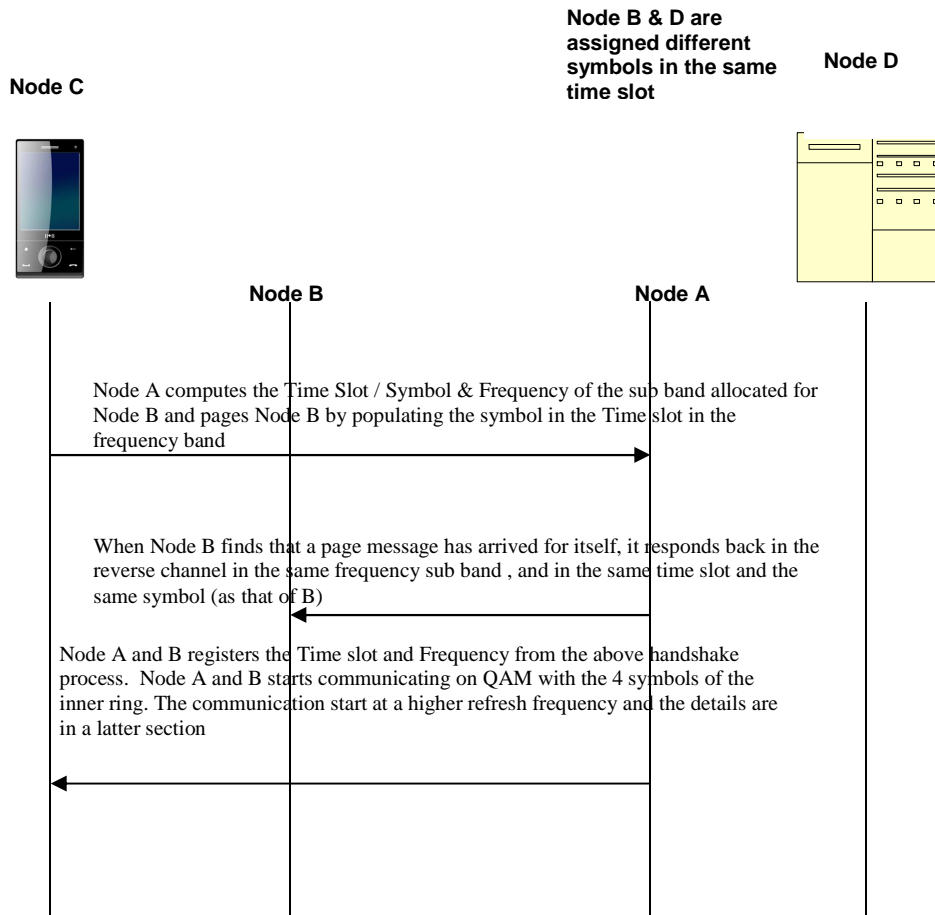


Figure 30 : Call flow – Inter cell Handover

3.2.3.Types of Handover and the handover criteria

The network area is divided into square shaped cells. Attempt should be made to do the cell allocation in such a way that they aligned laterally and vertically in the north south direction as shown in Fig 31.

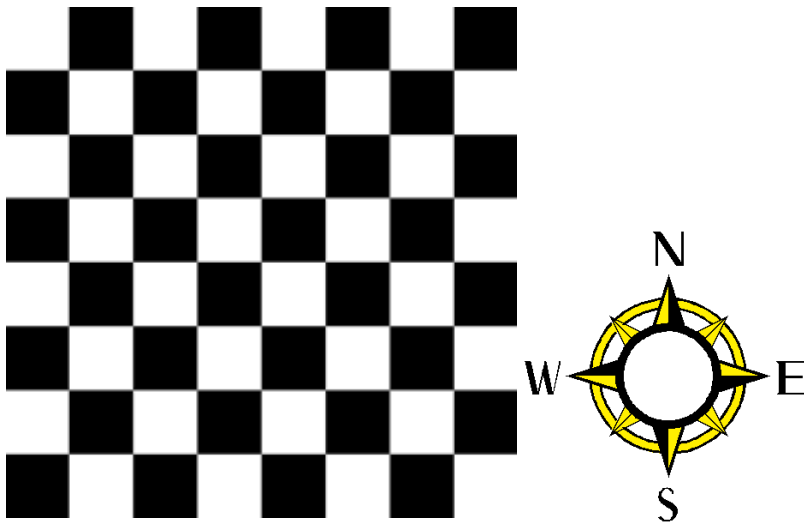


Figure 31: Checker board facing North South

There are 2 types of handover possible.

Lateral Handover: In this case the mobile station currently being served by a particular cell, moves to another cell with the opposite shade.

The MS monitors the direction of travel (either by Compass or GPS), calculates the rate of increase/ decrease signal strength from the 2 transceivers and then the MS chooses the right cell where handover can be initiated.

If the signal strength decreases below -75 DBm, as available from the active transceiver, the processor in the MS decides to initiate a handover to the adjacent cell.

In case of a lateral handover, it may happen that there are multiple pilots available from the adjacent cells. Hence the MS determines the direction of motion to derive the probability of entry in the appropriate cell, and also from the rate of increase/ decrease of the pilots of the available passive adjacent cells.

Diagonal Handovers: Actually there will be no direct diagonal handovers as shown in Fig 28. The passive part of the transceiver can only monitor the frequencies of the cells which have the frequencies in the adjacent block of the AFCRN. So handover always happens in the cell with the opposite colour.

If a MS is travelling exactly diagonally, then there will be two handovers. Initially, the handover is actuated towards the cell of the opposite colour (of the present cell). Subsequently, the handover happens to the cell with the same colour level.

So the diagonal handover means more processing required at the Mobile station

Interesting to note is that, it is the Mobile Station which initiates the handover and not the network. The intelligence and service logic is also intrinsic to the mobile station. The mobile station does not have any communication to the external network, except in transferring the present Time Slot, Cell ID and Frequency information to the new cell (Coordination processor). So the signalling related functionality / exchange, especially at the application layer is reduced to a considerable scale.

The max transmit power level of the MS is around +33 dBm Watts . As the size of the cell is close to that of a present day mobile network , hence we take +33 dBm as the maximum transmit power of, which is the standard [13].

The Max transmit power level of AP is 320 watt. All the cells in the network areas need to generate equal power level. All the Cells should be same in dimension as well. The following figures provide a rough estimate of the power levels during the different phase of the operation of the mobile station.

The figure 32 represents the amplitude of the MS transmission (maximum being +33 dbm) on reception of an incoming call and the various network operations during the process.

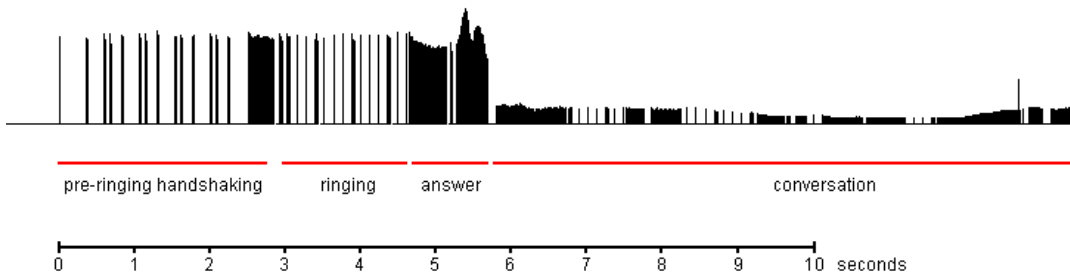


Figure 32: Power levels during the phases of operation in a call.

Because of silence suppression mechanisms, there are no data transmitted during the silence period.

3.3. Determination of the timing frequency for address symbol transmission and the data bit transmission.

As shown below, f_r is the sampling frequency applicable for the frame, while f_d denotes the symbol rate for data traffic.

This implies that the sampling frequency for the timeslots carrying the symbols for addressing a node in the outer circle in the complex plane is lower than that applicable for the inner circle of the complex plane.

Ideally f_r could have only depended on the Busy Hour Call Attempt (BHCA) of the network.

But it is not so, as the addressing and the data traffic are on the same channel and the frequencies (f_r , f_d)are interrelated.

The symbols pertain to the inner circle of the constellation diagram.

$$f_d = 1/t_d = 1/3.7 \mu s$$

$$t_r = 8 \text{ symbols (referred to as Burst in GSM) .}$$

t_r is a single time slot carrying symbols of the outer ring on APSK. After the traffic channel is established, the t_g shifts to traffic mode and then starts to carry the traffic data at frequency of f_d .

$$t_r = 156 * t_d$$

$$t_r = 577 \mu s$$

Considering a 26 Multi Frame arrangement, the total number of Time slot (t_g) that can be realised per sub channel frequency is $26 * 8 = 208$.

Buffering Speech and transmitting at bursts

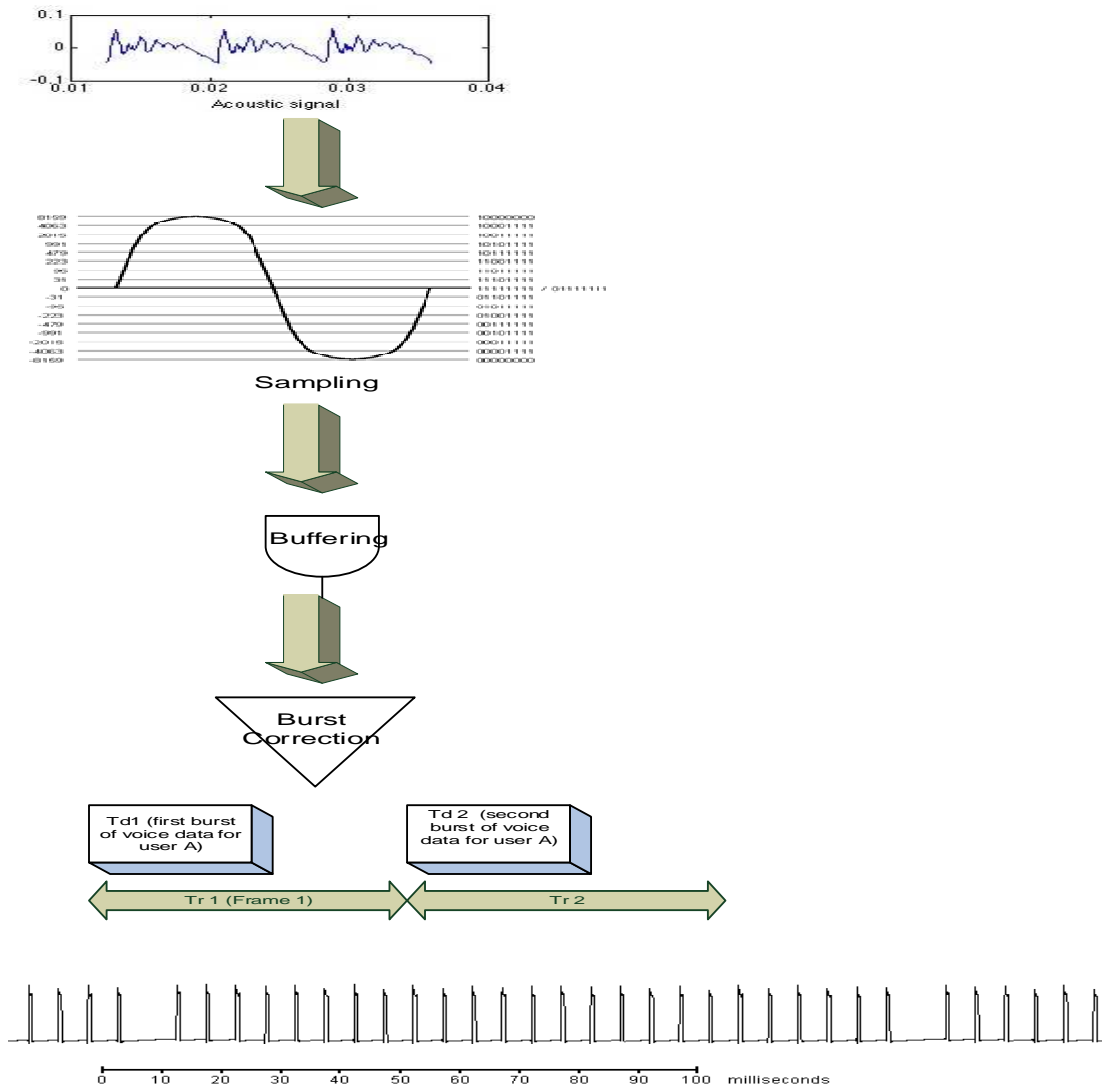


Figure 33: Burst formation on speech signal

Once the traffic channel is established after the addressing procedures are completed for identifying the Mobile Station in the network, the t_d timeslots (refer Fig 13) are used for transreceiving the symbols meant for the data traffic corresponding to the inner ring of the constellation diagram (Fig 3). The pulses come at 4.62 millisecond intervals (approx. 217Hz frequency), each lasting 0.57 milliseconds. This gives a mark: space ratio of 1:7, allowing up to eight calls to be time-multiplexed (TDMA) onto the same carrier frequency. Every 26th pulse is omitted, causing an 8.3Hz periodicity in the signal. Fig.33 demonstrates how voice is transmitted in bursts.

3.4. Comparison of the proposed Multiple Access Schemes

Below (in Fig. 34) is a synopsis of the evolution of the Multiple Access Schemes used by various technologies.

The constellation diagram derived from the various technologies is provided below. Starting from the basic GSM to HSDPA and LTE, we see that the fundamental principle of all these technologies at the lower layers remain the same. As the access technology evolves, it caters to more data speed by squeezing more symbols on the complex plane. Hence the distance between the symbols decreases and so are the chances of error. Hence newer methods of error corrections are required. The layers 5 to 7 become more intelligent to cater to advanced mobility management, location management, high bandwidth and presence related applications.

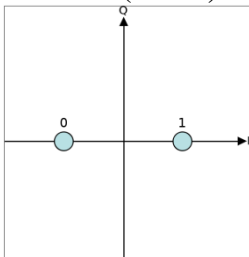
More intelligence in the application layer necessitates the devices and the network elements to be more complex, and hence increases the cost and the complexity of network processes, design and operations.

In figure 33, we see the constellation diagrams for each access technology (2G/3G/4G) and also the proposed technology. There is a fundamental and philosophical drift from the present day technologies, as we see in our proposed access scheme. Here we see that a certain category of the symbols which lies at the outer ring (propagated at very low frequency, proportional to the call attempt rate of a mobile network), are employed for identifying / addressing the user in the network plane. As the frequency is low, hence symbol error factor and loss is relatively decreased.

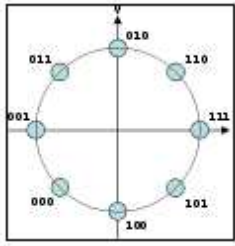
The other 4 symbols in the inner ring and propagated at the real data rate following QPSK, the same followed today by various mobile technologies.

So with this mechanism, the same time slot is used for control and data, with different multiple access schemes, once by 8 PSK for addressing and then with QPSK for data traffic.

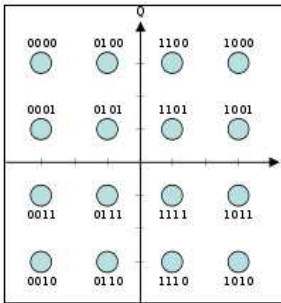
GSM 2G (BPSK)



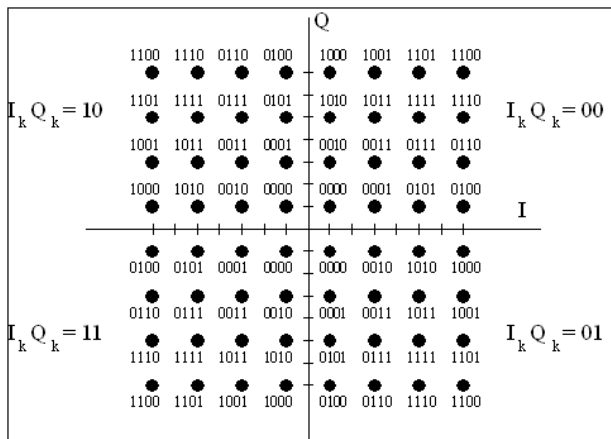
3G GSM –UMTS (8PSK)



LTE (16 QAM)



LTE-A (64 QAM)



Smart Multiple Access

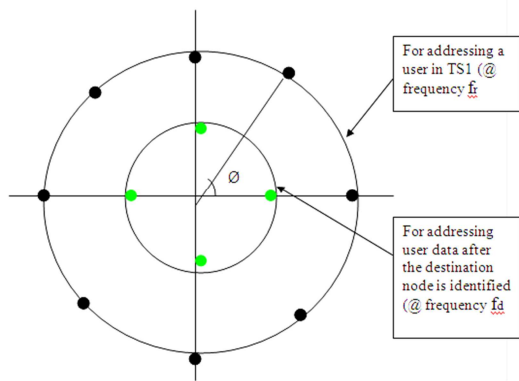


Figure 34 : 4 Constellation diagrams for 2G/3G/4G and proposed Smart Multiple Access

3.5. Comparison of the technologies

Proposed Multiple Access scheme	GSM/CDMA
Cell Planning: The entire network area is one big cell, so no cell planning required. The BTSs does not have any specific E.212 / CGI (Cell Global Identity) address as there are no cells. So the mobile station and the network do not need to keep a track of the BTS through which the Mobile Station latches on.	Cell Planning: Network Planning is a key activity for a cellular network. Proper cell planning is essential for quality of service. Cell planning requires substantial amount of efforts. The size, shape, number of cells, number of sectors and the frequencies to allocate for each cell and reuse are the main parameters.
Addressing: As the multiple access scheme facilitates to directly address a node at the physical layer, hence less signaling resources are required.	Addressing: This is done at the layer 7 by BSSAP in case of GSM. In case of CDMA, the methodologies are described in IS95/CDMA 2000 [6] or the protocols beyond.
Mobility Management: The system is agnostic to the exact location of the node in the network area, hence no complex procedures of mobility management followed	Mobility Management: Signaling processes, like Location Update (GSM) [8], Registration Notification (CDMA), different types of handoffs (hard, soft, Intra /Inter MSC, etc) are a part of the mobility management process and which requires lot of network resource and bandwidth.

Layer 7 activities of the Network devices: The network devices for the RN are involved in layer 7 activities only for supplementary services and call control	Layer 7 activities of the Network devices: The BS system needs to carry out the layer 7 activities for location & mobility management, addressing and call control.
---	--

Table 2: Comparison between proposed (Novel) scheme and GSM/CDMA.

3.6. Addressing and Numbering Plan followed for the global mobile network

The Mobile station will be assigned the E.212 address, while at the core network, this will be mapped to the E.164 or ENUM address, as is prevalent in modern day mobile communication systems. The HLR /HSS will hold this mapping. Hence in case of paging, the E.212 address will be used to address and identify the user in the network area. When the subscriber is provisioned in the HLR/HSS, the Symbol coordinate, Time stamp and the base frequency number will be mapped to the E.212 address of the subscriber. The E.164 address is the phone number of the Mobile station.

3.7. Comparative analysis of the signaling processes and overhead for mobility management in the State-of-the-Art and the proposed smart network

The unique mobility management procedures in SMNAT is not designed specifically for one specific application scenario but for supporting coexistence of M2M, M2H, H2M and H2H applications. The motivation is to reduce the signaling interaction between the network and the device, actuate a better congestion control and thereby alleviate the impact due to existence of heterogeneous devices in the mobile network. The following section to scrutinise the major differences between the State of Art and the SMNAT in terms of signaling and mobility management is applicable for all types of User Equipments including the M2M terminals.

3.8. Synopsis of Mobility and Location management in LTE

The signaling overhead due to mobility management is related to location update, paging, registration, handover, call origination and termination. During TA (Tracking area) update procedure [13], the Mobility Management Entity (MME) [7] records the TA in which the User Equipment (UE) is located. When a UE, i.e. the Mobile device moves to a new TA, tracking area update is initiated by the device. Paging comes in play when the UE is being called for voice, SMS, network initiated Location messages (for location services) and network initiated USSD. In order to place the call to the UE, MME broadcasts paging message in all cells of the UE's registered TA. In LTE, the Mobility Management Entity (MME) is responsible for the mobility management function. The MME is connected to a large number of evolved Node Bs (cells) that are grouped into the Tracking Areas (TAs). The TAs are further grouped into TA Lists (TALs). When a UE moves out of the current TAL, it reports its new location to the MME. If the LTE network attempts to connect to the UE, the MME asks the cells in the TAL to page the UE. In LTE paging, the MME may sequentially page a cell, the TA of the cell, and/or TAL of the cell.

In the following sections, we investigate the 3 main processes of a mobile network related to mobility and location management which imparts signaling overhead in the network. A comparative analysis is made between the State-of-the-Art and the SMNAT.

Location update: State-of-the-Art (LTE)

Location update in LTE happens can be categorised as following.

Static scenarios

- Always-update: In this scheme, the user updates its location whenever it moves into a new cell.
- Never-update: In this scheme, the user never updates its location, which means that the location update overhead is zero, but may lead to excessive paging.
- Reporting cells: In this scheme, the user updates its location only when visiting one of the predefined reporting cells.
- Forming LA: In this scheme, the user updates its location whenever it changes an LA.

Dynamic Update Schemes

- Selective LA update: In this scheme, the LAU is not performed every time the user crosses an LA border.
- Time-based: In this scheme, the user updates its location at constant time intervals.
- Profile-based: In this scheme, the network maintains a profile for each user. The profile has a sequential list of the most likely LAs that the user is located at different time periods.
- Movement-based: In this scheme, the user updates its location after a given number of boundary crossings to other cells in the network.
- Distance-based: In this scheme, the user updates its location when it has moved away a certain distance from the cell where it has last updated its location. This process is primarily based on Reference Signal Received Power (RSRP) calculations by the network.
- Predictive distance-based: In this scheme, the network determines the probability density function of the user's location based on location and speed reports.

The MME houses the location update algorithm to cater to any of the above mentioned scenarios. The signaling overhead for a location update message as seen in the core network is around 378 Bytes. Depending upon the scheme chosen, the total overhead for exchanging the Location update message can be determined. Fig. 35 shows a Location Update message showing layer 3 to layer 7.

71 Smart Mobile Network Access Topology

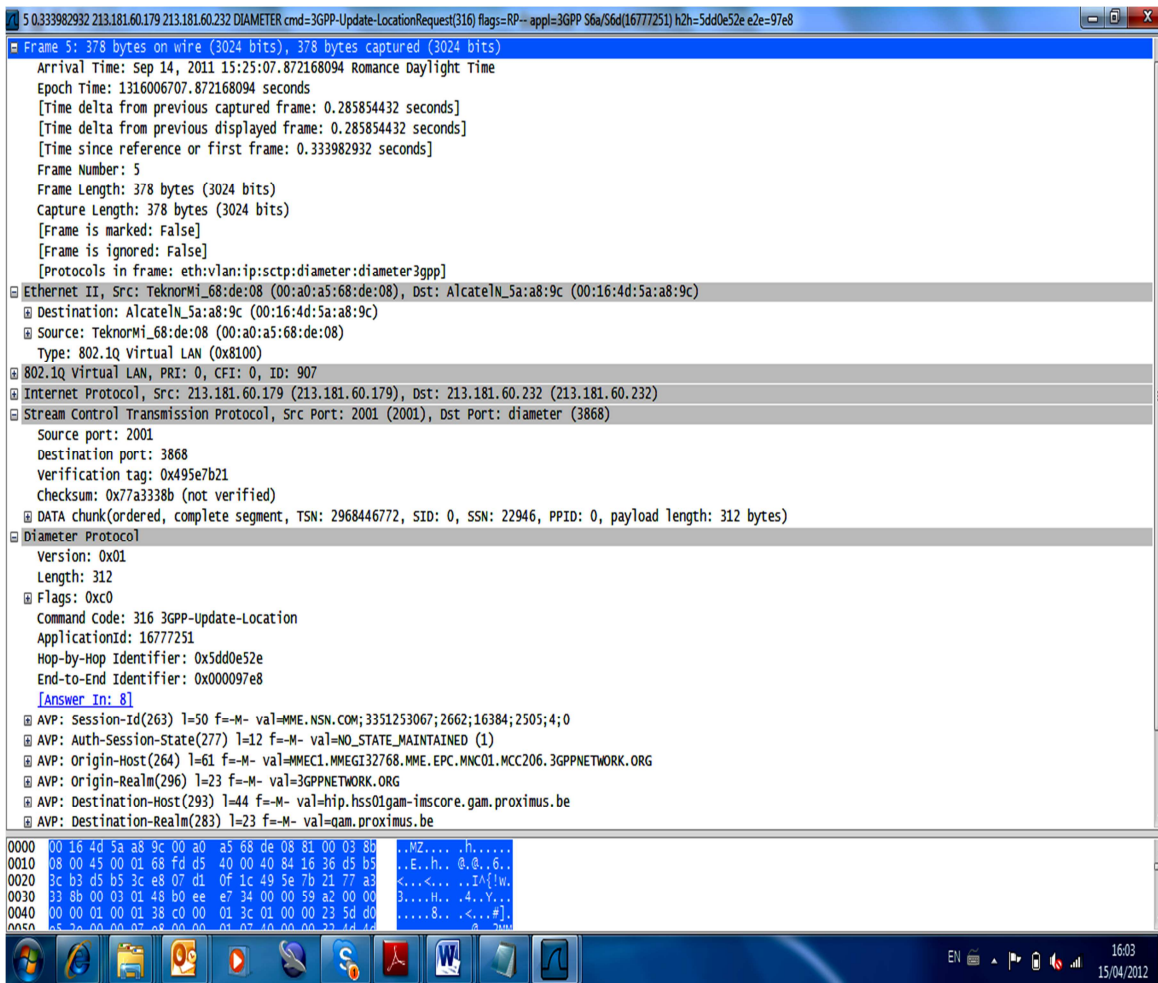


Figure 35: Update Location Request on Diameter S6 Interface

Before the application layer message for location update are sent, the LTE handset initiates the synchronisation and random access process to acquire a channel. Fig. 36 demonstrates the synchronisation and the random access process for the LTE access layer.

Synchronization Procedure

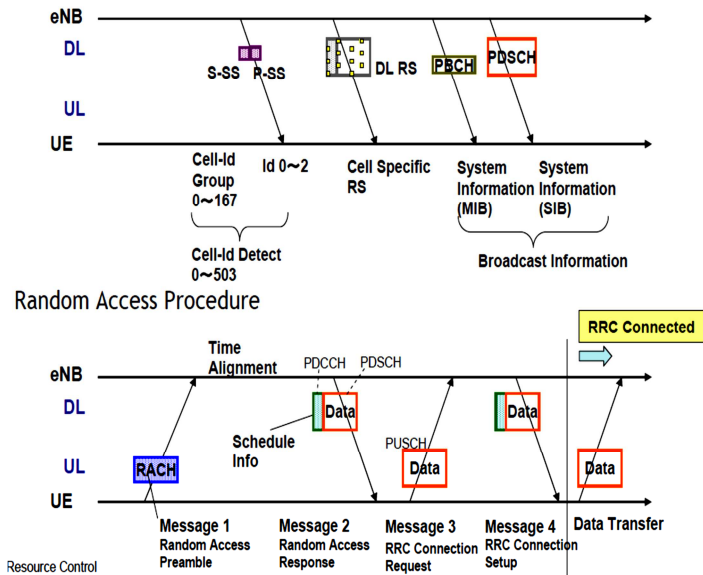


Figure 36: Synchronisation and the Random Access process for the LTE Access layer

The processing of MIB /SIBs reception by the handset (40 ms in case of MIB and 80 ms in case of SIB 1) consumes battery power in the handset.

The bandwidth required for Location update can be calculated as follows:

The set of cells in a network is denoted by $N = \{1, \dots, N\}$, and the set of TAs currently in use is denoted by $T = \{1, \dots, T\}$. The vector $\mathbf{t} = [t1, \dots, tN]$ is used as a general notation of cell-to-TA assignment, where ti is the TA of cell i . TA design \mathbf{t} can be alternatively represented by an $N \times N$ symmetric and binary matrix $\mathbf{S}(\mathbf{t})$; in which element $s_{ij}(\mathbf{t})$ represents whether or not two cells are in the same TA, i.e.,

$$s_{ij}(\mathbf{t}) = \begin{cases} 1 & \text{if } t_i = t_j, \\ 0 & \text{otherwise.} \end{cases} \quad [6]$$

h_{ij} is the number of UEs moving from cell i to cell j

The overhead (Bup) of one location update

$$B_{up} = \sum_{i \in \mathcal{N}} \sum_{j \in \mathcal{N}: j \neq i} (c^u h_{ij} (1 - s_{ij}(\mathbf{t}))) \quad [7]$$

3.9. Location update in SMNAT

There is no location update process as there is no network acquisition required in SMNAT. More importantly, there is no location update required prior to the execution of the other network operations / processes, for example while receiving calls/SMS or during handovers. For these operations the necessary precondition is successful paging of the UE at physical layer but the UE does not need to attach to the given network. Thus by avoiding location updates, we save substantial battery power of the UE as well as spectrum bandwidth.

Paging in LTE

In most cases, this paging process happens while UE is in idle mode. This means that UE has to monitor whether the networking is invoking any paging message to it and it has to spend some energy (battery) to run this "Monitoring" process.

The paging procedure is used by the network to request the establishment of a NAS signaling connection to the UE. The network shall initiate the paging procedure for EPS services using S-TMSI with CN domain indicator set to "PS" when NAS signaling messages are sent to the UE.

Paging in LTE happens via PDCCH channel. The UE listens to the PDDCH channel in given intervals, but not continuously. This is done by DRX (Discontinuous Reception) in order to save power.

The DRX LTE has two timing units as many of other technology. Timing Unit in Frame scale (SFN: System Frame Number) is one unit and the timing unit in sub frame level (Sub-frame Number). It means that the network needs to know both SFN and Sub frame Number to locate exact position in LTE time domain. Regarding the paging cycle, PF (Paging Frame) + PO(Paging Occasion) let you know the exact timing when UE has to wake up to catch the paging message being sent to it.

$$PF = SFN \bmod T = (T \div N) \times (UE_ID \bmod N) \quad [8]$$

According to 3GPP 34.104, T is defined as follows:

T is DRX cycle of the UE. T is determined by the shortest of the UE specific DRX value, if allocated by upper layers, and a default DRX value broadcast in system information. If UE specific DRX is not configured by upper layers, the default value is applied.

It means UE can get the T from two different sources, one from the system information (SIB2, IE defaultPagingCycle) [30] and the one from upper layer. If upper layer send the value, it use the T value from the upper layer, otherwise UE has to use the value from SIB2.If T is 5 Milliseconds, then the handset will not be able to receive a page message unless the DRX sleep cycle is completed.

Paging in LTE is done at PDCCH using the S-IMSI (SAE- Temporary IMSI) for addressing which is a 32 bit value.

3.9.1. Paging in SMNAT

The address for the page message is the symbol coordinate at a specific time slot in the radio frame (in the Predefined AFCRN) assigned to the UE. The UE is always in the sleep mode, except the fact that it has to wake up to listen to the specific time slot assigned to itself in the available AFCRNs available within the spectrum. The UE does not need to listen

to the entire radio frame. This is not possible in the State-of-the-Art as the paging information can arrive dynamically from any time slot in the frame.

The time slot carries only a single symbol with a repeated transmission of 4 times. If each symbol measures a time span of 1 micro second, the total time of the radio frame is 10 ms (say), then the UE has to remain live for 4 micro seconds. Lower time to live implies better battery power conservation.

Secondly, no higher layer addressing parameters are used for paging which also conserves the bandwidth. The layer 7 parameters are exchanged only when the paging process is completed and the time slot is seized between the UE and the CP (coordination processor) for the conveyance of the signaling and data information. This is the primary advantage of the SMNAT over the state-of-the-art where it is not possible to confine the paging process to a layer 1 activity.

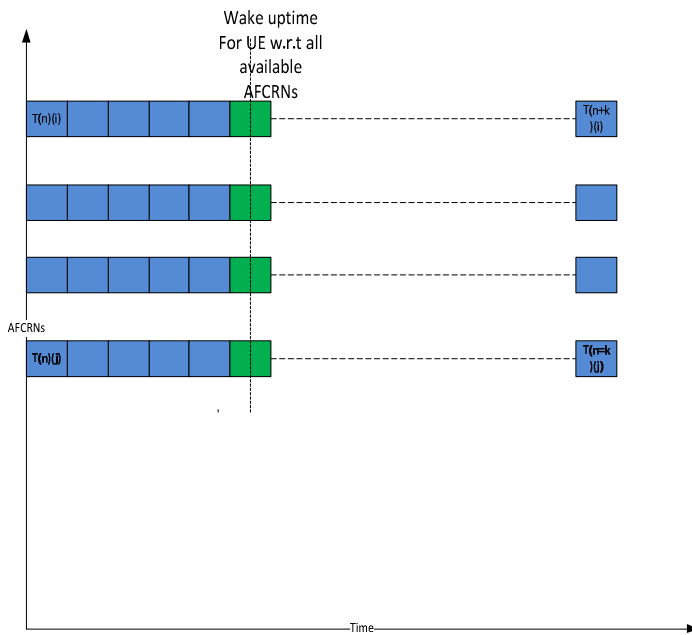


Figure 37: Wake up time for the UE

The time period during which the UE needs to wake up can be calculated as follows (refer Fig. 37) Let each frame pertaining to a single AFCRN contain n to $n+k$ time slots.

Say the coordinate symbol (of the complex plane) assigned to the specific UE arrives in K^{th} time slot. Let there be AFCRN's ranging from i to j .

Hence the time period for which a UE needs to be awake within the radio frame T_{live}

$$T_{live} = 4 \sum_{i=0}^j t(n) \quad [10]$$

where n can range from 0 to K .

As we have a repetitive code (4 repetitions) for error correction, so the total time is multiplied by 4.

3.10. SMNAT as enabler of Device to Device communication

The 5th and 6th generation of mobile networks are envisioned to realize the bandwidth required for WWW (Wireless Worldwide Wide Web). They should be highly dynamic in nature and actuate self- optimization of resources to support the bandwidth for ever increasing customer base. Spectrum crunch is one of the major issues. Hence these networks should thrive to achieve very high spectral efficiency and less Carrier to Interference ratio (C/I), relative to the existing technologies like LTE and WIMAX. Data speed per user should also not be compromised. Ubiquitous, instantaneous and always connected are some of the key prerequisites of a modern mobile device. It has been partly complied by LTE and some Network based applications based on the IMS (IP Multimedia Subsystem) [46] framework like Rich Communication Suite (RCS)[47]. The data requirement per user is facing an exponential growth with the high proliferation of the applications on the UEs available today. Machine to Machine applications (M2M) and Internet of Things (IoTs) today play an important role in shaping up the data traffic patterns which can be sporadic or distributed over time. As the new device technologies usher in, the data throughput requirement per user soars. The smartphone and tablets with 3 Dimension (3D) screens and cameras will be bandwidth hungry. Screen resolution will further increase leading to higher data rates. Online multiplayer gaming applications running on 5G gaming consoles with 3D displays will require higher throughput both for the uplink and downlink. Ultra High Definition (UHD) voice and UHD video realised through Rich Communication Services (RCS) [47] will contribute to the growth of data traffic. More dedicated data bearers will be required at the access and core to guarantee the Quality of Service (QoS) promised to the user. Due to the swarm of Machine to Machine (M2M) and Internet of Things (IoT) devices in the network, the mobility management related processes will be more intense and will consume more network resources. The nano nature of the devices with limited RF power will imply additional small-cells/ microcells which will increase the intricacy of the access and core network.

3.10.1. 5G communication landscape

With the present day device capabilities, the mobile devices are in constant interaction with each other. The human centric Mobile device interacts with various wearable devices, like smart watches, wearable computers (eg. Sixth Sense), SOS devices and health equipments. As the user moves around, these devices become more dynamic in nature. The devices, categorised as M2M or IoTs may be as below (though this is an example and not an exhaustive list:

1. Environment sensors
2. Connected cars
3. Smart objects and robots
4. Health equipments
5. Small cells not owned by the mobile operators.

According to the newly formed group called METIS (Mobile and wireless communications Enablers for the Twenty-twenty Information Society), which is a consortium of mobile OEMs, operators, academic institutions and automotive companies a 5G Network should have 1000 times higher mobile data rate volume per area, 10 to 100 times more connected devices, 10 to 100 times higher typical user data rate, 10 times longer battery life for low power devices and 5 times smaller end to end latency. The main objective of the project is to lay the foundation of 5G, the next generation mobile and wireless communications system. The aim is to let people seamlessly bridge the virtual and physical worlds offering the same level of all-senses, context-based, rich communication experience over fixed and wireless networks. Apart from these basic needs, one need systems which can work with the same performance level in the crowds. It should

offer the same QoS and throughput as in office, or home as on the move. It should render low end to end latency and reliability to enable machine type applications.

Standards around 5G are evolving. So an accurate definition of 5G still does not exist. Below is a snapshot of how some network vendors visualise 5G.

5G mobile communications technology is the next generation of 4G LTE networks technology and can offer data transmission speeds of up to several tens of GBPS per base station. Once 5G networks are commercialized, they will allow users to transmit massive data files, including UHD movies and remote medical services, “practically without limitation,” Samsung claims. A 5G network will comprise of small cells densely clustered together to give a continuous coverage implementation of Dynamic Spectrum Access.

According to Huawei, Massive capacity for delivery of services available in 5G will allow connections between end users and the network to be made at “faster than thought” speeds – so fast that the apparent distance between connected people and connected machines will shrink to a virtual “zero distance” gap. An instant immediacy in mobile services will lay the foundation for a whole new set of mobile apps to proliferate and push the capabilities of communications beyond what is currently possible. A more massive capacity for managing connections will better enable a greater widespread adoption of M2M services and interactions, and will facilitate innovation in localized mobile service delivery.

The connection essentially comprises “short” wireless links on the end of local fiber optic cable. It would be more a “nomadic” service (like WiFi) rather than a wide area “mobile” service using the millimeter wave bands (20 – 60 GHz) so as to allow very wide bandwidth radio channels able to support data access speeds of up to 10 Gb/s.

3.10.2. Related work on 5G

To meet these demands and to conceive a network which incorporates all these enhancements over the 4G, one may need to analyse the potential of new access technologies and architectural concepts and then identify the ones which can be the eligible contenders to serve as 5G technology

Some of them are as follows.

- a) Multi-Tier 5G Networks
- b) Physical Layer based Network Coding
- c) Generalized frequency division multiplexing (GFDM)[55]
- d) Non Orthogonal Multiple Access (NOMA)
- e) Cognitive Radio
- f) Device to Device (D2D) communication
- g) Beam Forming technology
- h) Millimeter wave technology
- i) Massive MIMO

However, the scope of this chapter is not to investigate on all these options and not to provide a comparative analysis. Rather, some new paradigms which have the potential to play pivotal roles in shaping up the 5G network architecture and the D2D communication framework are staged. In D2D communication, user equipments (UEs) exchange information amongst themselves peer to peer over a direct link using the cellular resources instead of through the base station or eNodeB. This is markedly different from small cell (femtocell) communication where UEs communicate with the help of small low-power cellular base stations. D2D users communicate directly while remaining controlled under the cellular access network. This optimises resource utilisation in a cellular network and boosts spectral efficiency.

This chapter will primarily delve in the following domains.

- a. D2D communication: a study on the existing approaches with the new approach, the Smart Mobile Network Access Topology (SMNAT).
- b. Integration of SMNAT with LTE-Advanced and 5G Core.
- c. Security aspects in light of cooperative communication between 2 devices.

3.10.3. Cellular device to device communication

Cellular Device to Device communication is meant to reduce the cellular traffic load by actuating a breakout from the UE itself and directly establish traffic towards the other paired UE using the cellular channel. As a collateral impact, this can significantly contribute to slacken the processor intensive signalling process as explained above. It leverages the benefit from the proximity between two devices and increases the overall resource utilisation of the cellular network. But it is imperative that one needs to come up with new methodologies for device discovery and pairing. Direct D2D technologies have already been developed in several wireless standards, aiming to meet the needs for efficient local data transmission required by variant services in personal, public and industrial areas. Some of the existing contenders are Bluetooth, Zigbee and direct WiFi. With D2D communication, the aim is to find a method which is tightly integrated with the cellular network and uses the same spectrum as cellular operations.

Imparting D2D capability in a mobile device impacts the whole of the network framework and is not an inconsequential addition. Issues like authentication, real time billing, fraud control will crop up and the devices directly will interact with each other bypassing the network.

But on a positive note, the benefits are bountiful.

1. As the paired devices are in the same cell sector, or in the same cell (different sector) or in adjacent cells are in physical proximity, high data rates with low latency can be achieved.
2. Depending on the proximity of the devices, the radio power level will be reduced. This will result in better battery life.
3. As the same radio resources are used for cellular and D2D communication, hence the average frequency reuse factor will be better.
4. For a traditional cellular network, one needs 2 distinct channels for uplink and downlink between the UE and the BS. But in case of D2D communication, a single channel can be used for both directions. Hence the overall spectral efficiency is better.
5. The spectral efficiency can be further enhanced if the cognitive radio communication is used wherein the unused spectrum holes may be utilised for establishing direct communication between the 2 nodes.
6. As the D2D communications has limited dependence on the network infrastructure the devices could be used for instant communications between a number of devices within a range.
7. D2D on 5G would use licensed spectrum and this would enable the frequencies to be used to be less impacted by interference.
8. In times of natural calamities where some essential components of core and access network have failed, D2D communication can ensue.

As a practical implementation scenario of an Over the Top (OTT) application meant from smartphones (for device to device communication) is from Google. A new mobile messaging application called FireChat is empowering nearby smartphone users to stay in touch even when there's no cellular service or Internet connection. The messaging app harnesses a technology called wireless mesh networking, which might someday allow a myriad of devices to connect like links in a chain. The technique might someday be used to tie together thousands of devices with built-in radios and make it possible to be online without having to pay for the access. It could also enable online communications in remote areas or disaster zones without Wi-Fi or cellular signals.

D2D has been proposed as a Rel.12 3GPP feature. D2D Study Item had an approval in 3GPP SA1 (Services working group) in 2011, called ProSe (Proximity based Services)[48] which identifies the use cases and envisage the requirements including network operator control, security, Authentication, Authorisation and Accounting (AAA) , regulatory aspects, public safety requirements, integration with current infrastructure, network offloading. The ongoing discussion by ProSe includes evaluation requirements, D2D channel model, resource use, ProSe discovery and ProSe communication, etc.

ProSe Communication between two UEs in proximity is established by means of a communication path established between the UEs.

- The ProSe Communication path is established
 - By Direct communication between the UEs
 - Or routed via the local eNB.
 - By ProSe Discovery
- Communication Process identifies that a UE is in proximity of another UE.
 - by Open [ProSe] Discovery
- ProSe Discovery without permission from the UE being discovered.
 - by Restricted [ProSe] discovery
 - ProSe Discovery that only takes place with permission from the UE being discovered.

Device to Device communication for M2M type devices is a topic of interest for many telecom researchers. This is because of the huge volume of the devices which may eventually clog up the Mobile network and jeopardise the human to human services. The study on D2D requirements for MTC Device to MTC Device scenarios covers

- The identification and functionality needed to set up a connection towards a MTC Device.
- The IMS domain may provide a solution for this required functionality.
- MTC Devices often act as a gateway for a MTC capillary network of other MTC Devices or non-3GPP devices

D2D group for Machine type communications (M2M) Study on Enhancements for MTC is a 3GPP specification - TR 22.888 [49] has been crafted in purview of these requirements.

As depicted in Fig 38, the scenarios that are covered are

- Devices Communicating directly
- Devices Communicating via MTC Server
- Devices Communicating with assistance from a Name resolution server

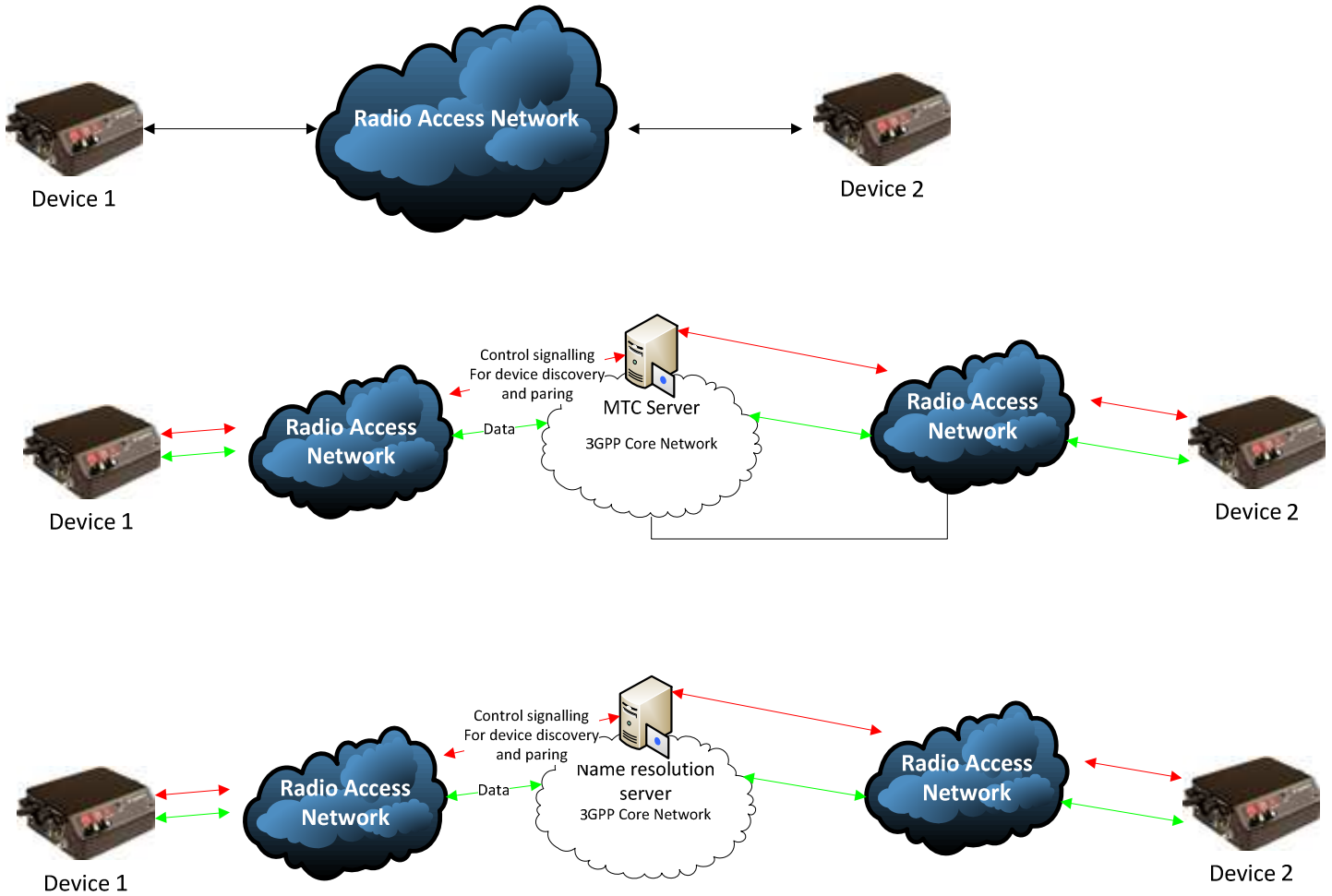


Figure 38: Three methods of D2D communication for M2M devices

D2D Communication can be broadly categorised as:

1. Out of band
2. In-band

Out of band D2D essentially implies that the devices use a radio technology in the standalone mode to actuate communication with a paired device. The pairing method is in the purview of that specific radio technology.

Some existing out of band D2D communication method are:

- Bluetooth
- ZigBee
- Near Field Communication
- Direct Wifi

In-band D2D communication implies that the devices use the cellular spectrum and the cellular technology.

In this chapter, a new concept for the realisation of in-band D2D not by using new resource allocation methodologies but to use a new mobility management and addressing concept called SMNAT (Smart Mobile Network Access Topology) [13] has been presented.

The subsequent sections will bring out the various radio resource management technologies that can be employed in the in band D2D communication, as seen in the State-of-the-Art. These technologies are based on the underlay cellular network, hence they fall in the category of in-band D2D communication methods.

3.10.4. D2D using physical layer network coding

In this technique [51], there are two cooperating mobile nodes which can relay network codes over the channel codes in 2 different paths, as shown in Fig. 39. Firstly it is being sent to the candidate mobile, and subsequently it is being sent to the BS. The candidate mobile node responds back with a new code. The code received from the candidate mobile is combined with the code that was originally sent to that mobile. The combined code is sent to the BS which modifies it with a mathematical expression and relays to the candidate mobile. The candidate mobile who is aware of this mathematical expression deciphers the code that it had sent/received to/from the mobile that has initiated the pairing. When it is successful, the channel is identified and the traffic path between the two nodes is established.

The random selection of 2 pairs may not alleviate the performance of the network and optimise the system capacity. This is because of the fact that channel conditions between the mobile node and the base station can vary to quite some extent. This may lead to disproportionate SINRs between the nodes and the BS, entirely depending on the channel quality. Hence User Grouping based on proportionate parameters is done by the network based on some Cost Function so as to optimise the performance.

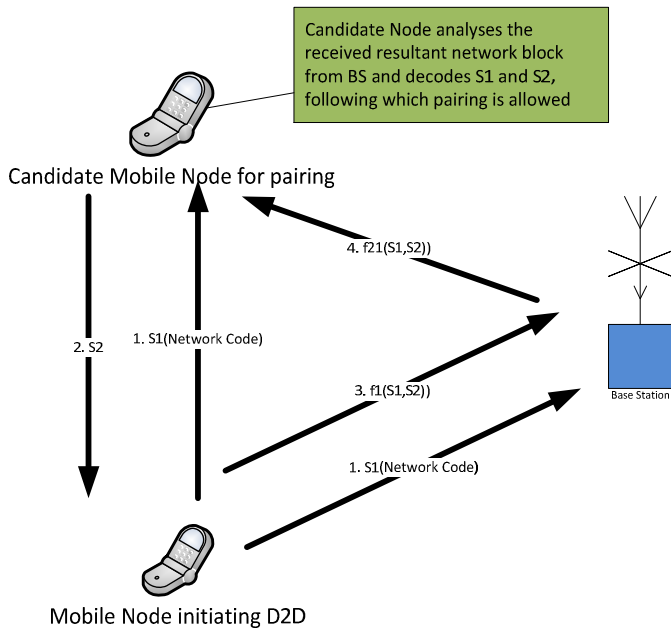


Figure 39: Interaction between the Initiating Device, candidate (pair) device and Base Station for realising Network Assisted D2D communication

3.10.5. D2D using Fractional Frequency Reuse (FFR)

Interference is a significant issue in D2D communications. The interference is caused by devices in D2D mode with that in the normal cellular mode and the eNodeBs. It also depends a lot on the position of the D2D device in the cell, whether it is towards the exterior or towards the interior.

In [52], the radio resource allocation scheme using FFR is proposed. Different resources are allocated to the D2D UEs according to their location in the cell. If the D2D UE resides in the inner region of a cell, then they can use the frequency band that is not used by the eNodeB for relaying to the D2D UEs [55]. The ratio of different frequency reuse factor and the corresponding power level are optimized or adjusted adaptively according to the traffic load and user distribution. D2D and eNodeB relaying UEs located in the same cell do not interfere because radio resources from another frequency band are orthogonally allocated to the D2D and the eNodeBs relaying to the UEs.

3.10.6. D2D using cognitive radio

In [53], there is an overview on how D2D communication can be realized over secondary users. With this concept, the primary users can only transceive via the BS. The secondary use can avail both D2D plus the BS transmission mode.

In [54] a joint subcarrier and power allocation method CR-D2D-MC for cognitive multicast with D2D communication coexisting with cellular networks has been proposed. The impact of imperfect spectrum sensing is considered in the proposed problem, which results in the capacity decrease of the cognitive multicast. The simulation results show that the proposed algorithm improves the spectrum efficiency and maintain a better trade-off between capacity and fairness for cognitive networks in a low algorithm complexity. Therefore, employing cognitive multicast based on D2D is able to explore more potential spectrum resources adequately to improve the system performance, and make it possible to satisfy the requirements of multiple kinds of high rate transmission.

3.10.7. Integration of SMNAT with 5G core

It is imperative that the success behind any new technologies lies in its interoperability and integrability with the existing or evolving networks. As a reference, in [45], the primary challenges for interworking between LTE core network and legacy core network are discussed. SMNAT is oriented on the physical layer to implement the processes of addressing, mobility management and data exchange. On the contrary, 3GPP's vision on mobility management on 5G is founded on the application layer, more specifically the NAS (Non Access Stratum). The NAS defines the basic processes for mobility management for EPC (Enhanced Packet Core) between the UE and the MME.

We follow a two stepped approach to integrate SMNAT with 5G network as elaborated subsequently..

Short term approach

WLAN can be integrated by the EPC core as it permits integration with non 3GPP untrusted network via the ePDG (evolved Packet Data Gateway). From the architectural perspective, WLAN and SMAT have some proximity as both of them have similar network topologies. Hence as shown in Fig 40, the end to end integration between SMNAT and EPC is shown.

The UE establishes a communication with HSS through ePDG for actuating EPC authentication. At SMNAT end, RADIUS based authentication is followed alike the EAP-AKA mechanism. The conversion rules from RADIUS to DIAMETER (EPC) follow the GSM IR 61 recommendations. The data transfer between the UE and the IMS core is established via the S2b interface between the ePDG and the PGW.

The call flow is shown in Fig 41. It can be seen that 2 processes are covered.

1. Authentication
2. Data transfer

It may be interesting to note that mobility management messages are not exchanged with EPC. In a State-of-the-Art (LTE) network, the UE actuate NAS signalling with the EPC. NAS, which is a layer 7 process is not implemented in SMNAT. Rather SMNAT is dependent on a process where location management is realised via layer 1. This is the reason, the mobility management messages like Location Update / Location Cancellation are not seen (Fig 41). SMNAT can directly fit in the architecture which has been realised for integrating WiFi with EPC.

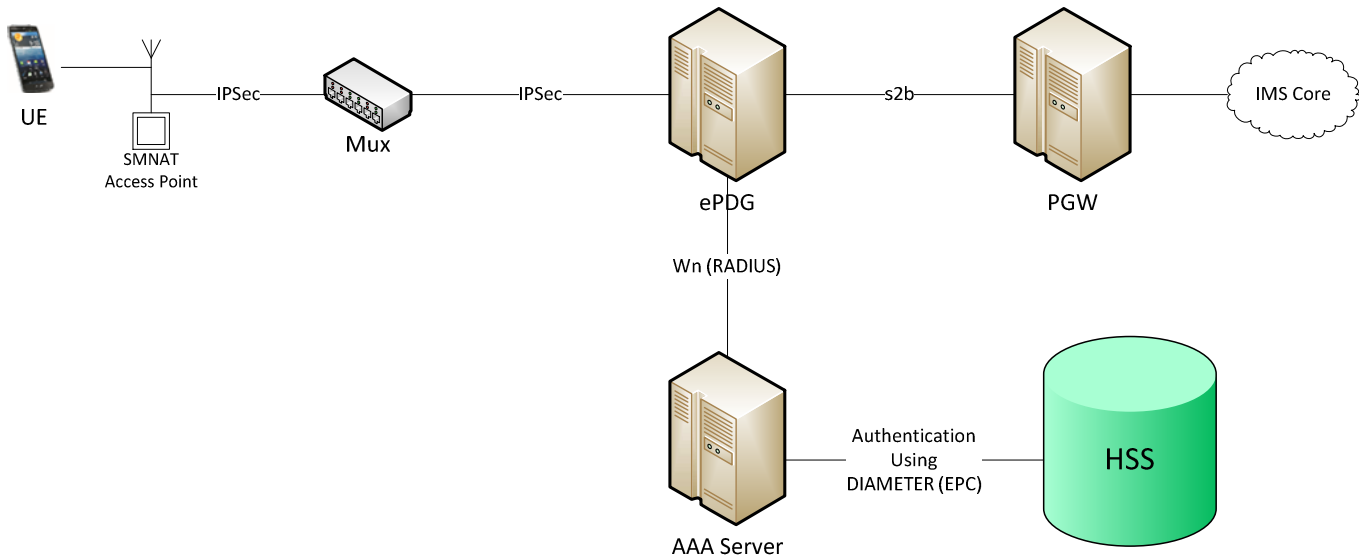


Figure 40 : Integration of SMNAT with EPC

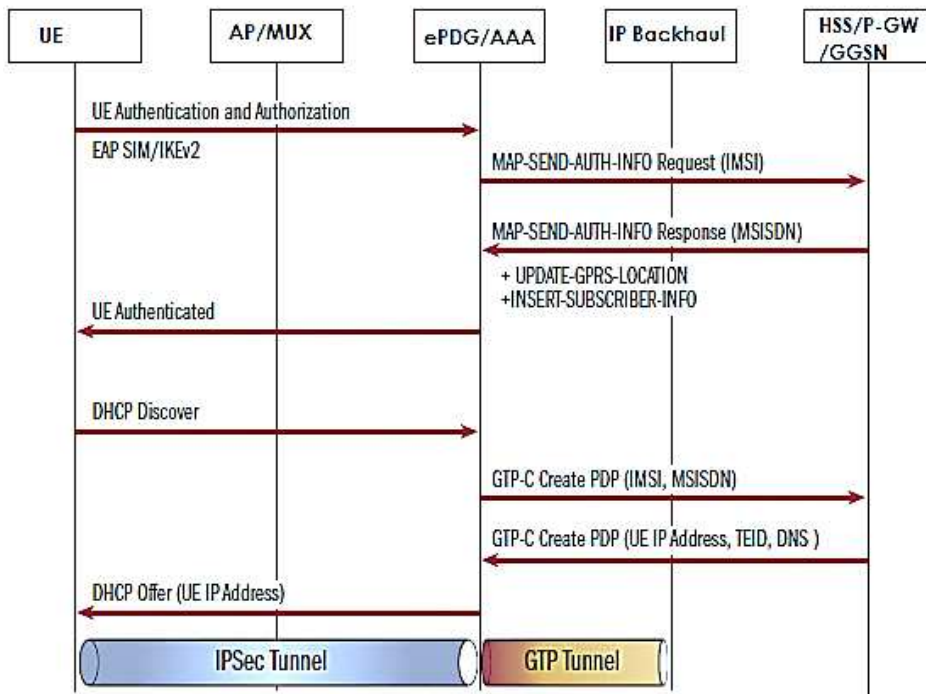


Figure 41: Call Flow between UE, SMNAT access and EPC core

Long term approach

The long term approach is to evolve the eNodeBs and MMEs as to be compatible with SMNAT architecture. As the mobility management part is simplified, these smart eNodeBs will thrive on physical layer addressing, rather than the communication on NAS. The MMEs will be closer to the definition done for Coordination Processor, which can directly engage itself in physical layer addressing and mobility management. Hence both eNodeBs / Home eNodeB and MMEs will be lighter in terms of power consumption and processor capacity than the state-of-the-art networks.

3.11. Implementation of SMNAT for in-band D2D and interoperability with WISDOM

This section brings out how SMNAT can be used to realise device to device communication, which can solve some issues and achieve some specific objects which the State-of-the-Art D2D technologies cannot attain. Contrary to the work done so far on D2D where the peering is only possible between the 2 devices in physical proximity, SMNAT offers a more versatile solution where the D2D leg also could be established between 2 UEs which are not in vicinity. The following section discusses how this can be achieved.

SMNAT can be deployed as a access and core network layer (Fig 42), located on the same network plane as the WISDOM 5G [57][58][59] based on cognitive communication. In such a scenario, SMNAT will be dedicated for D2D communication, while WISDOM will be responsible for global cellular roaming within the network area or beyond.

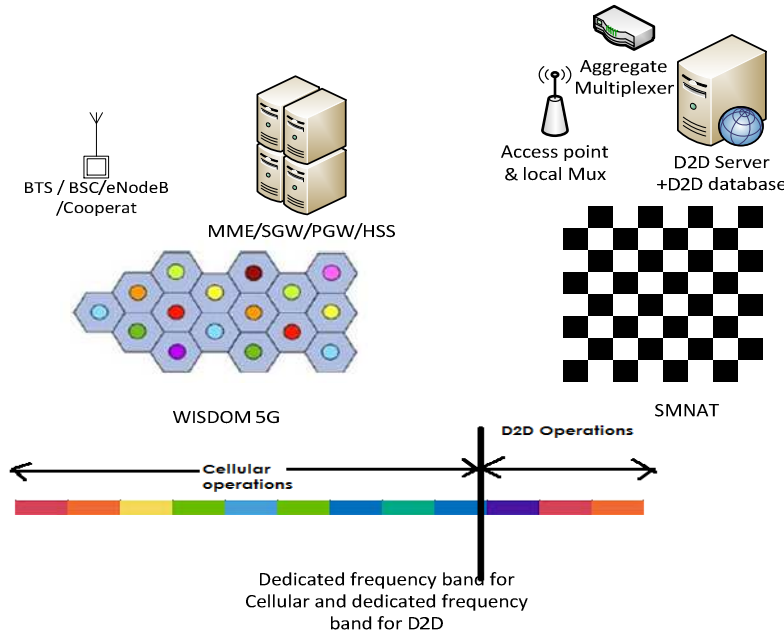


Figure 42: Coexistence of WISDOM 5G and SMNAT

The available spectrum will be allocated between SMNAT and WISDOM. The ratio of the spectrum allocation can be determined through the process of network planning and can vary according to the traffic characteristics and the business requirement of the particular operator.

To decide whether D2D leg will be established or not, there are 2 possible options.

1. Decision by the user.
2. Decision by the network based on cooperative communication

When any device initiates a communication process on data, it will have 2 options. It will be given a chance to try a D2D communication with the incentive of a faster communication process or lesser data charges (if offered by the Network Operator).

If the device does not opt for the D2D option, then the session will be established via the normal process of Enhanced Packet Core (EPC) signalling. Subsequently, the EPC can further decide whether to terminate the session to the candidate device via LTE / LTE-A, or the network will send instructions to the device rather to automatically initiate a D2D session.

3.12. Description of network elements of SMNAT and the call flow for session establishment.

SMNAT [13] [1] has a unique design of the cellular network which represents a checkerboard. There are access points in each cell, which via the Frame Aggregator can communicate with the D2D server. The D2D server is interfaced to a D2D database which holds the mapping between the E.164 address of the mobile device and the SMNAT coordinates, namely the symbol coordinate, time slot and AFCRN.

As evident from Fig 43. D2D transport channel can be established directly between 2 nodes in case they are in proximity (same or different cell), but facilitated by the SMNAT Access and core network (for signalling). In case the devices are not in the direct signal range of each other, then the traffic channel will be established via the SMNAT Access and core (Access Point \leftrightarrow Frame Aggregator \leftrightarrow D2D Server). But this process uses the addressing and mobility management methods pertaining to the physical layer as defined by SMNAT.

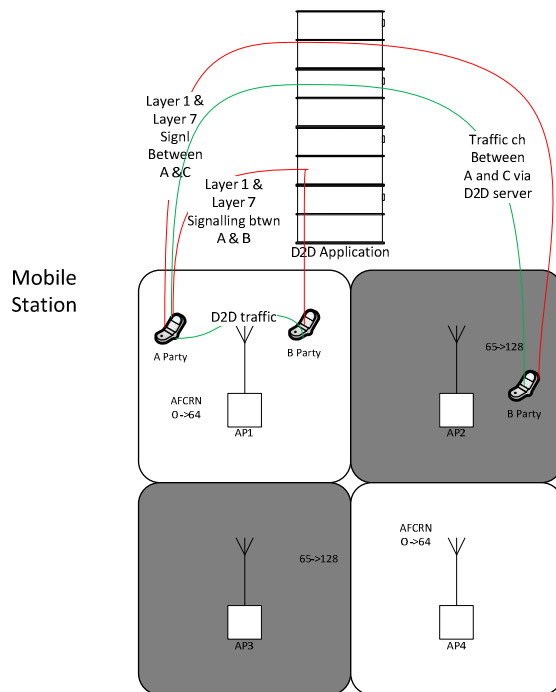


Figure 43: D2D Channel establishment based on proximity of the paired device.

In Fig. 44 the end to end call flow for D2D channel establishment is shown, which is self-explanatory. In this scenario, the Mobile device A originates a call towards mobile device B where there are two possibilities.

- The devices A and B are in the same cell. In this case the channel is established via local access point in the vicinity of the devices.
- The devices A and B are in different cells. In this case the channel is established via the D2D server.

D2D server is a core network application platform which is responsible to establishing the pair, either directly between the two devices or via the core network using physical layer addressing and mobility management.

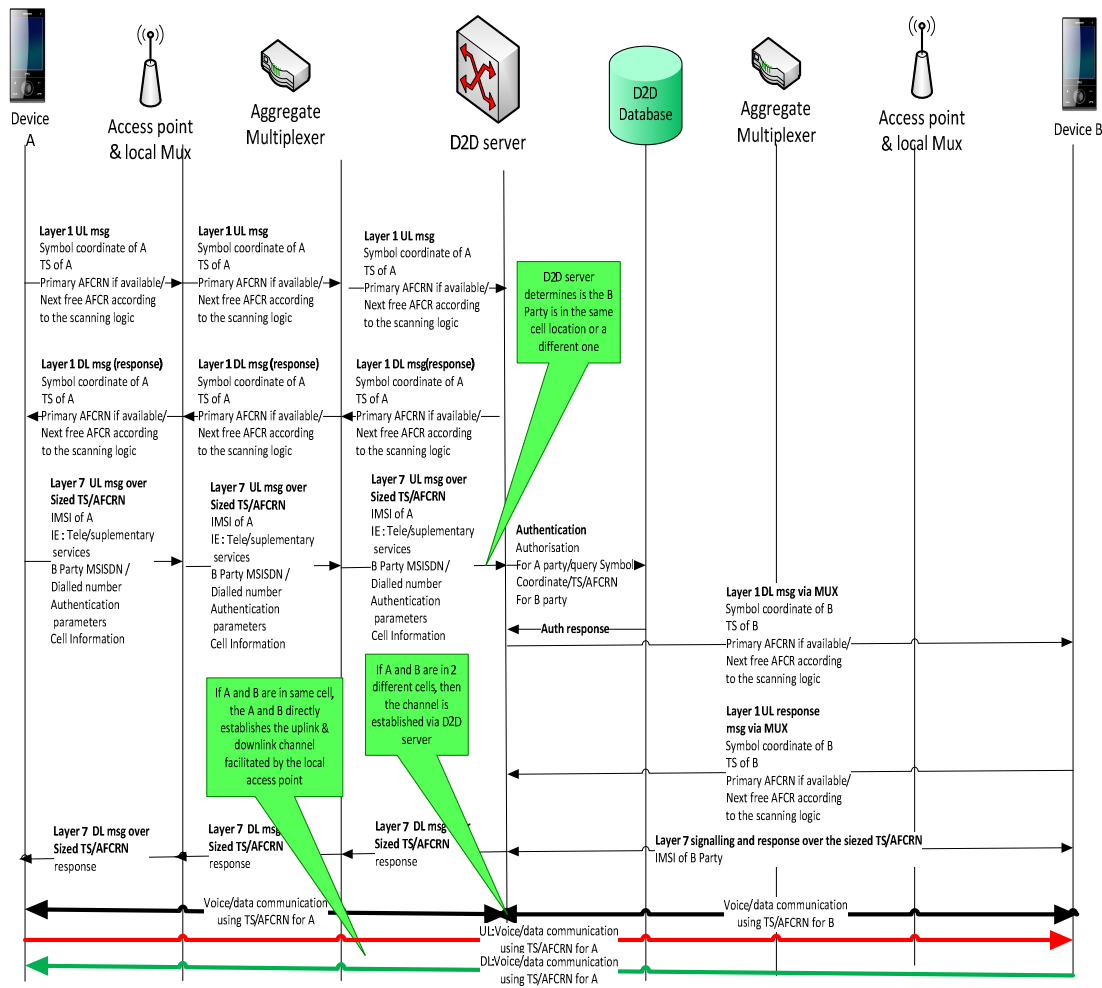


Figure 44: End 2 End communication channel establishment for D2D for a Mobile Originating , Mobile terminating call.

3.13. Decision by the network to initiate D2D leg based on cooperative communication

In case the user attempts (by default) to establish communication through BS via the 5G network [56][57][58] network, then it will first check if the session can be terminated by the D2D channel. This implies that the network will check if it can offload from cellular to D2D for capacity and optimisation reasons. This can be done with the help of Diameter interaction between the HSS and the D2D server. A high level process is shown in Fig.45.

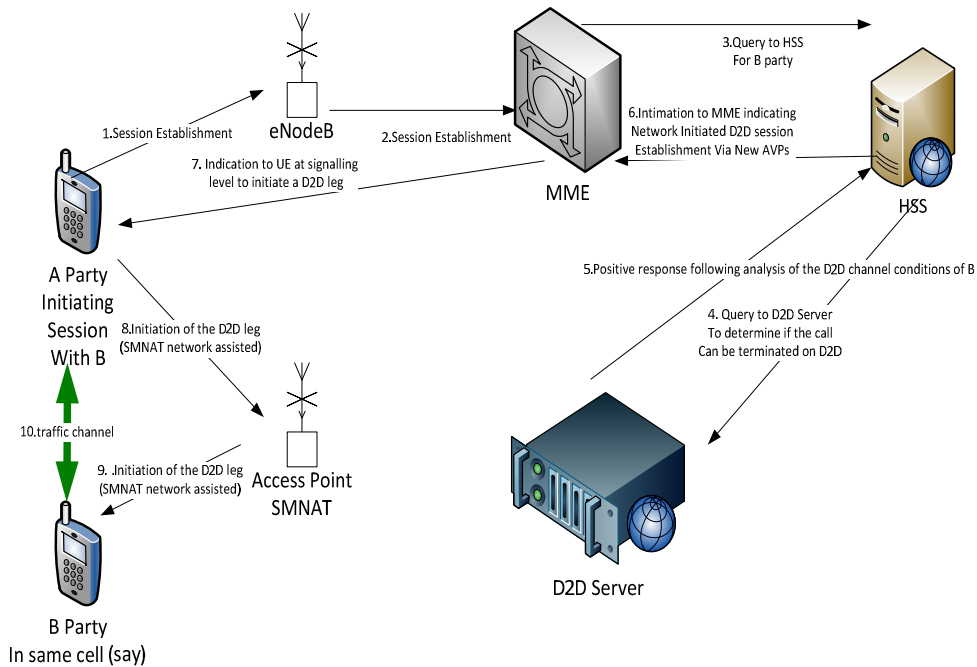


Figure 45: Network Initiated D2D channel establishment

3.14. Security aspects in light of cooperative communication between 2 devices in cellular D2D mode

As explained before, D2D communication can be established in the direct mode or via the network depending upon the proximity of the devices, channel conditions and location (both devices in the same cell or different cells). Network Security have become a primary concern especially for the D2D communication.

In case of the network assisted D2D communication, the network can enforce the security methods that are in place for cellular communication. The integration of SMNAT with the EPC has been shown in Fig 40 & Fig.41, where the conventional EAP AKA method meant for traditional WLAN network can be used. The ePDG (Evolved Packet Data Gateway) /AAA (Authentication Authorisation Accounting) platform modifies the RADIUS based authentication messages to AKA based EPC

authentication parameters over DIAMETER directly with the HSS. The EPC authentication vectors including ciphering keys that are downloaded from the HSS will be retranslated and compared for the purpose of authenticating the UE.

However, for the direct mode of D2D communication, where two devices need to pair with each other using the cellular channel, the procedures become intricate. Generation of symmetric keys for the communicating peers without key exchange is challenging. One of the methods is to generate the keys is based random variations of wireless channel like Channel State Information (CSI). Most of these methods usually use the measurement results of individual subcarriers. But this is not robust, given the fact that the subcarriers in proximity can have similar channel conditions and have strong correlation dependence. Due to this fact there can be repeated segments in the key, which may be easy to crack. Another approach is to use RSS Key generation mechanism based on channel measurements. But the main issue is that it is not designed to meet the requirements of a system demanding key generation at massive scale. Bit generation rate is low as a single sample can provide a single RSS value. This also becomes vulnerable if the UE is static and the channel conditions are predictable and the channel variations are predictable. Hence the endeavour of the researchers is to realise a method of key generation based on CSI, but not based on a single subcarrier. In [59], a new algorithm termed as KEEP has been proposed. KEEP aims to actuate the key generation process by incorporating the following sub-processes.

1. Dropping the inconsistent bits by exploiting the correlation of CSI measurements from multiple carriers. This is executed by a federated filtration method.
2. Implementation of Universal key for validating consistency of bit streams between two communicating channels. A pair of devices exchanges only a part of the hashed keys, which makes it difficult for the attacker to guess and formulate the whole key.
3. Using methods of key recombination and adaptive quantisation to generate secret bit from a large number of subcarriers. The bit mismatch rate with the process is low. Moreover, it eliminates the correlation of bit streams from multiple subcarriers and reduce predictivity.

The has key generation process based adaptive quantisation procedure, CSI sub – blocking and hashing is captured in the following algorithm in Fig 46.

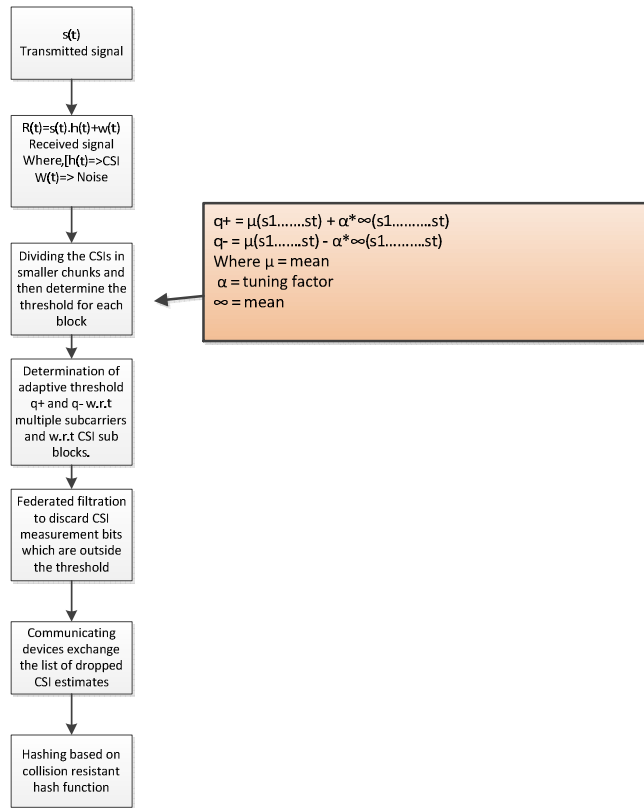


Figure 46 : Key Generation process

3.15. Comparison between the D2D approaches related to radio resource allocation management (network coding, fractional frequency reuse) and SMNAT.

Scenario	Network Fractional Reuse	Coding/ Frequency	SMNAT
Channel establishment based on the proximity of the mobile nodes	Traffic offload only for cases where the 2 nodes are in proximity		In case the nodes are not in proximity, D2D can still be established via the Access points / D2D server based on physical layer. So no additional layer 7 activities are involved after the traffic channel is established
Spectrum allocation	Do not need dedicated spectrum		Needs dedicated spectrum
Initiation of the service	Service can be only requested by the user		Can be initiated both by network and user

Interference	Prone to interference as the frequency is shared with cellular usage	No interference issue, as spectrum is dedicated
Integration with network	It is an evolution of the radio resource management, but not a separate solution	It is a standalone solution which can be integrated with WISDOM 5G
Optimisation of the cellular network	D2D leg can be established when it is initiated by the user and when the nodes are in proximity. So cellular traffic offload can be done only for these specific cases	There will be more frequent cases of D2D traffic breakout because : 1.D2D leg can be initiated not only by user , but also by network 2. D2D leg can be established also when the 2 devices are not in vicinity and out of network range of each other

3.15.1. Summary and conclusion (SMNAT for D2D)

At present, mobility management is explicitly in the domain of the application layer. This chapter introduces SMNAT where it is proposed to shift from this paradigm and knit this functionality with the physical layer. This entails simplification of network architecture and associated processes and helps in attaining a network which has a degree of location and presence agnosticism. The interaction between the device and the network occurs only when one of them initiate a network operation for a call, SMS or other services. The handover operation is also simplified. Further, a novel Device to Device communication concept based on SMNAT has been proposed. Unlike the existing researches in D2D, engrossed in redesigning the radio resource allocation process, a parallel network stratum has been proposed which can actuate D2D communication based on addressing and mobility management processes closer to the physical layer. The processing cycles in the network will diminish which creates an extra room to cater new customers.

5

Channel Impairments and error control mechanisms

4.0. Introduction

A signal propagating through a wireless channel is subject to various physical effects of radio wave propagation.

Some physical effects impacting the propagation of the signal are provided below.

- Path loss and attenuation on obstacles
- Reflection, diffraction, scattering
- Interference (adjacent or co-channel)
- Thermal or man-made noise
- Imperfections of transmit / receive circuitry

The proposed multiple access scheme is not free from these physical effects and the losses due to them. But the impact on the addressing of the mobile nodes due to these losses may be a bit magnified compared to the present day mobile access technologies.

Let us explain why. With the present mobile communication technologies, like GSM/UMTS, the access channels and the traffic channels are distinct and separate. There are dedicated error correction mechanisms defined for these two categories of the channels.

For the existing technologies the mobility management related functions are dependent on the higher layer functionalities. So while there is a basic error correction at layer 2 to ensure that the correct information (symbols) are carried over to the far end, but then it is mainly the layer 5 to 7 which actively take part in actuating the mobility/location management and the addressing related functions of the mobile station. At layer 5/7, there is an extra leg of validation and authentication carried out to ensure that the addressing information has arrived for the correct mobile node.

In our proposed technology, the addressing part is directly handled by the lower layers, namely layer 2 of the OSI stack. The errors in the physical channel can distort the coordinate of the symbols in the complex plane, leading to wrong addressing. Hence any error at access channel implies that the addressing related functionality is impacted, which means the rate of success of any call events like call maturation, SMS, data session setup etc is jeopardised. So it is extremely important to ensure that we employ the correct and a robust error correction mechanism to ward off this kind of situation, which can in effect endanger the practicality of such a multiple access technology. The major difference between the proposed technology w.r.t the GSM/CDMA technologies is related to the impact of error control. The details are as follows.

As indicated that we do not have 2 separate channels for the conveyance of signalling and data information. Instead a time slot is first used for the addressing purpose, and when the process is completed and the target node identified, the same time slot transitions to a state identified by the mobile node, so that the time slot can be used for data communication.

Hence we propose two different error correction mechanisms, one to be used when the time slot acts as an addressing channel, and a different error correction mechanism when the mobile node acts as a traffic channel.

4.1. Proposed Error Correction Mechanism for the Time Slot Acting as the Addressing Channel.

In the outer circle, there are a total of 8 symbols per time slot meant for addressing the mobile nodes. When a transmitter (either a MS or AP) transmits a symbol to the receiver it will transmit odd number of times the same symbol. The frequency of transmission of this symbol is f_p which is determined according to the Call Attempt per second in a mobile network. Hence the frequency is low (compared to the frequency of traffic data propagation) and hence is less prone to error.

We propose a 2 step process to the error correction mechanism. The process starts with a repetitive code as explained. In such codes each bit of the message is repeatedly sent an odd number of times so that a simple majority decode rule can be followed to recover the original message. For example, the repetition code may send each symbol three times and then decide to decode and determine the given symbol. So in general, if each bit is repeated $2*n + 1$ times then the code can tolerate up to n errors.

Repetition codes are not very efficient as they increase the size of a given message by a factor of 3 or 5 say, depending on the repetition parameter. The redundancy added by a repetition code is just additional copies of the message itself, whereas more sophisticated codes add a smaller more intelligent amount of redundancy that can specifically pin point errors and correct them. More compact and intelligent forms of redundancy come at a cost of additional message processing for both the sender and the receiver. A definite advantage of repetition codes is simplicity both in encoding, and particularly decoding. So during any call processing when there is an attempt by the network to reach the mobile node, the network will transmit the same symbol pertaining to the specific coordinate in the complex plane and the fixed time slot (allocated for the target mobile node).

The receiver (mobile node) will calculate the error between the symbols by calculating the Euclidean distance between the reference symbol and the received one. If the mean error is less than Δe (error), then it will respond back to the network. So if the error is less than Δe for each incoming code, the target mobile node responds back with the same symbol repeated the same number of times. Below (Fig 47) is the handshake process between the network and the mobile station.

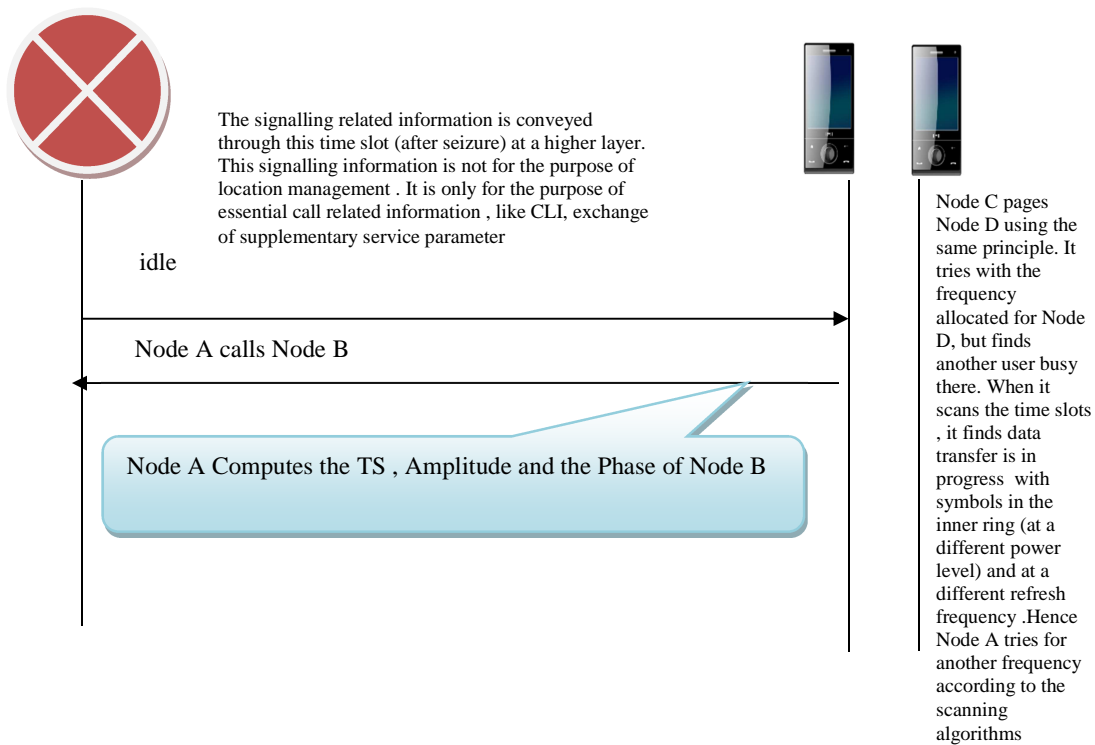


Figure 47: Error Correction by Repetitive Code

Now after the mobile node completes responding back with the same symbol $2n+1$ times, the Network again generate the symbol $2n+1$ times, to enable the MS to actuate a quick BER test to assess the channel quality.

For the addressing part, the symbols will be transmitted at the rate of the call attempts made per second (CAPS) in the network. Needless to say that the CAPS rate is much less compared to the actual traffic data rate. Hence due to the lower data rate, the symbols that will be transmitted following 8PSK modulation scheme (outer ring) will be much less prone to channel impairments compared to the symbols transmitted for the data traffic (inner ring). More the data rate, more symbols are packed in the complex plane implying that the spatial distance between the symbols decreases. This makes the channel more susceptible to error.

Hence we assume that 10^{-6} is a modest threshold for BER rate above which the error coding mechanism needs to be altered for the purpose of accuracy in the addressing symbol bit transmission and reception.

If the BER is 10^{-6} to 10^{-10} , then there is no further coding action is required and the network / mobile node can conclude that the identification process is completed. The mobile node and the network registers this process as completed for the given activity and related to the particular mobile node. However if the BER is more than 10^{-6} , then the mobile node generates a layer 7 identification message with the IMSI parameter written. It is transmitted in the same time slot towards the network.

The network has to respond to this message with the information regarding the call request (with the appropriate parameters). This response message shall be accepted and processed by the mobile node. The mobile node and the network thus reserve the time slot for the transition towards a traffic channel. The algorithm that will be used is below in Fig 48.

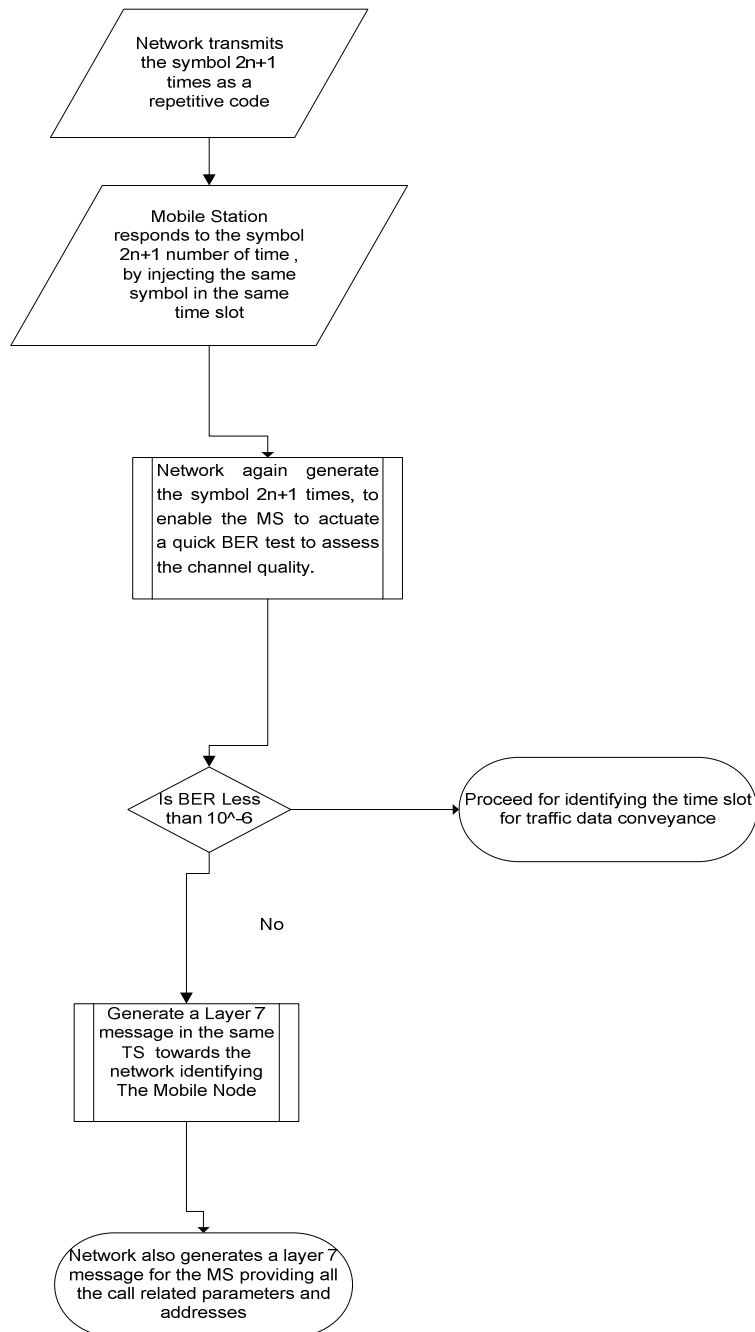


Figure 48 : Layer 1 and Layer 7 Error correction algorithm for the network and the Mobile Stations

During the generation of the Layer 7 identification message, the error coding scheme that should apply is generic involving the block code, convolution code and interleaving as witnessed in the state-of-the-art.

Each channel uses the following sequence and order of operations:

- the information bits are coded with a systematic block code, building words of information + parity bits;
- these information + parity bits are encoded with a convolutional code, building the coded bits;
- reordering and interleaving the coded bits, and adding a stealing flag, gives the interleaved bits.
- All these operations are made block by block, the size of which depends on the time slots available per frame.

4.2. Calculation for Symbol Error (Ps) for 8 PSK with SNR = 15 Db.

Symbol error and bit error will play a major role in the proposed scheme due to the fact that the addressing mechanism is dependent on layer 1. Though the chances of error is reduced by the fact that the sweep frequency for the frame comprising of TS td (section 2.1) is proportional to the Call Attempt rate , where the frequency is in order of Khz. In this section we derive an estimate of the probability of symbol error (as in equation 8). In general, we see that it stays in the permissible limit. Hence the network / mobile station do not in most of the cases of normal operation in good coverage areas will need to initiate the layer 7 error correction mechanisms as described in Fig.

We denote the coefficients $\{s_{ij}\}$ as a vector $\mathbf{s}_i = (s_{i1}, \dots, s_{iN})$ which is called the **signal constellation point** corresponding to the signal $s_i(t)$. The **signal constellation** consists of all constellation points $\{\mathbf{s}_1, \dots, \mathbf{s}_M\}$

The signal constellation for MPSK has

$$\begin{aligned} s_{i1} &= \text{Acos}[2\pi(i-1)/M] \\ s_{i2} &= \text{Asin}[2\pi(i-1)/M] \end{aligned} \quad [11]$$

$M]$ for $i = 1, \dots, M$.

The symbol energy is $E_s = A^2$, where A is the amplitude

Hence SNR for symbol is $\gamma_s = E_s/N_0 = A^2/N_0 \dots (N_0 \Rightarrow \text{Noise})$

$$\gamma_b \approx \gamma_s / \log_2 M \quad [12]$$

Let $\gamma_b = 15 \text{ dB}$, where , $\gamma_b = \text{SNR for bits}$.

SNR of 15 Db is considered , which is apparently high. But as the symbol rate is low for the addressing part (because it depends on the CAPS which is fairly smaller compared to the symbol rate for traffic data) , channel impairment is relatively lower. Hence SNR can be assumed to be on the higher side.

For the addressing symbols of the outer ring of the constellation diagram, we follow 8PSK.

Hence for 8PSK, $\gamma_s = (\log_2 8) \cdot 10^{(15/10)} = 94.87$.

Substituting this value into above equation 28 in Appendix 1,

$$P_s \approx 2Q\sqrt{189.74 \sin(\pi/8)} = 1.355 \cdot 10^{-7}.$$

$$P_b = 4.52 \cdot 10^{-8}. \text{ (substituting and approximating using equation 12)}$$

4.3. Proposed channel coding mechanism for the traffic channel.

Channel coding consists of adding to the source data some redundant information calculated from this source information. Decoding makes use of this redundancy to detect the presence of errors or estimate the most probable emitted bits given the received bits. Errors are detected when the transmitted redundancy is different from the one calculated with the received data.

Depending on the transmission mode, radio path transmission uses different codes. The codes that will be appropriate for the new multiple access technology are as follows.

- Block plus Convolutional Codes: these codes are only used for correction purposes. They achieve tremendous efficiency when they are combined with likelihood estimation such as that coming from the demodulator.
- Interleaving: Interleaving is the processes of rearranging the bits. Interleaving allows the error correction algorithms to correct more of the errors that could have occurred during transmission. It is combined with FEC codes in order to improve the performance of the error correction mechanisms. Interleaving decreases the possibility of losing whole bursts during the transmission, by dispersing the errors. Since the errors become less concentrated, it is then easier to correct them.

In our proposed multiple access methodology the signaling and the traffic information are based on the same timeframe. The timeslot which is used for signaling initially is reused for traffic channel. The sampling rate for the channels differ greatly after the signaling channel migrates to be a traffic channel once the signaling activity is completed.

So while the time slot was previously used for addressing and identification of the mobile node, the symbol rate was equal to the call attempts per second for the mobile network.

While the same channel is used for the purpose of exchanging the traffic data, the symbol rate is equal to the traffic data rate which is much higher. Hence the chances of the physical errors increase.

Important to mention is that the mobile nodes at both ends are aware about the state of the channels (signaling or traffic) and accordingly chooses the right coding process for achieving the required reliability.

We use the scheme followed by GSM for half rate and full rate channels [9]. The reasons for abiding by the GSM norms is as follows.

1. Quality tested and used by billions of users worldwide.
2. Existing SW/Firmware design employed for GSM can be re-used by this technology.

In brief, the following error correction steps are undertaken on the speech channel.

- Parity and tailing for a speech frame
- Convolution encoder
- Interleaving
- Mapping on a burst

Assuming that we have a 244 bit Enhanced full rate Speech Frame, the following Channel coding model comes in play [9]. This is shown in Fig 49.

The channel coding model is alike the principle followed by GSM/UMTS networks on the TCH frame for EFR speech frame.

As per Fig 49, the 244 bit speech frame is subjected to cyclic and repetition code. The output of the frame after adding redundancy is 260 bits. Depending upon the implementation at the operator end, half rate or full rate encoding is adopted. Subsequently, the time frame is subjected to cyclic code and convolution codes. The time frame is latter on reordered, partitioned and interleaved to obtain the encrypted output.

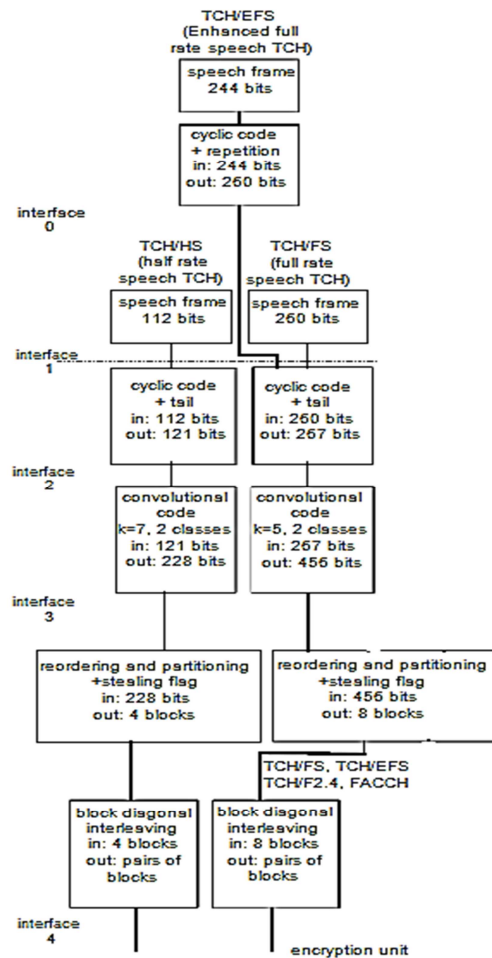


Figure 49 : The proposed code sequence scheme after the timeslot in the particular AFCRN band is chosen for traffic

4.4. Conclusion

It can be inferred that with the employment of the appropriate error coding scheme specifically (and separately) designed for addressing and data conveyance, it is possible to realize a robust and reliable overall access scheme. The addressing symbol correction is dependent on simple repetitive coding scheme if the BER is less. If it is high (assuming more than 10^{-6}), layer 7 coding is dynamically actuated.

For the traffic channel, the coding process is near to the process that we see for the traffic channels for the UMTS networks, as it is robust and presently used by millions of users worldwide.

6

Application of the Multiple Access Technology in the Wireless Local Loop

5.0. Introduction

Wireless technologies have been deployed in different parts of the world to provide local loop telecommunication services. Compared to traditional wireline (copper) local loop, Wireless Local Loop (WLL) technology solutions can be deployed much faster and at lower cost. Wireless technologies in the local loop can be broadly categorized as cordless (low-range), mobility (macro cellular) or proprietary. Cordless-based technologies are basically intended for high-density urban areas due to their short range. Since they require extensive backhaul infrastructure outlays, they have not been very successful in the present-day marketplace.

Mobility-based WLL are traditionally positioned for the macro cellular environment, which are well suited for rural areas. They are also used extensively for providing umbrella coverage in urban and suburban scenarios. Proprietary technologies are typically country specific and are intended for service within urban and suburban areas. Many of these solutions are however “Near Line of Sight”, thus limiting mobility and range. Fig 50 shows a very high level WLL deployment. The Access point, ie the Base Station Antenna is near to the mobile nodes. The mobility is limited to the locality covered by the Access point.

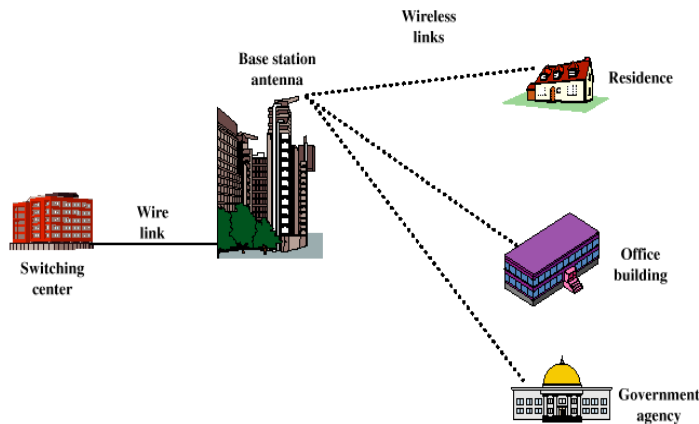


Figure 50: WLL implementation with limited mobility

5.1. Present Applications/Implementation of WLL

The term LMDS stands for Local Multipoint Distribution Service. LMDS uses microwave signals to transmit voice, video, and data signals using low power which can reach distances no greater than a five mile range. It is a wireless broadband service that relies on microwave radios to send large amounts of information between each of the radio at LMDS Hubs. One of the radios is installed at the LMDS Hub or station and the other is installed at the customer's site. The Hub or station is located at a key position to interface directly with land line fiber optic backbones as well as standard telephony.

Some key points for MDS service are

- Referred to as wireless cable for last mile access
- Used mainly by residential subscribers and small businesses
- Appeals to larger companies with greater bandwidth demands

5.2. Proposal for employing the proposed Multiple Access technology for Wireless Local Loop implementations

Colour in its electronic form, synthesized by the existing video technology, is generally used for the purpose of broadcast, multimedia and image processing. There can, however, be a more wide and positive use of colour if it can be employed in communications technology to serve as a carrier. In the pursuit for attaining higher data rates, the wireless technologies come closer to the Shannon Bound of the permissible S/N ratio, which is why, it is time to look for technologies that go beyond the conventional multiplexing methodologies for access layer (namely CDMA, TDMA, FDMA OFDM) and the available core network topologies. One such is to employ 'Colour Pixel Multiplexing' for the access and core layer for WLL. The colour pixel is used for addressing the wireless nodes in the network and for carrying the signaling and the bearer traffic. The present day video systems that can

generate millions of colours, in its electronic form have been utilized for setting up a wireless network, serving mobile stations or computers as its nodes. This chapter attempts to unfurl this new technique applicable for next generation WLL networks.

In the proposed network architecture, we propose a new network element called the colour server , which processes the video frame in the forward and the reverse channels, while each pixel of the video frames carry the voice and data information for the mobile users in either directions. It performs the task of a switching center that we see in the present generation WLL networks. The following section brings out the proposed network architecture in the light of the proposed Multiple Access Scheme.

5.3. Colour Pixel multiplexing in Access layer for wireless devices for the WLL network.

In a video frame, the number of pixels supported depends on the resolution of the frame. A pixel is generally thought of as the smallest complete sample of an image. The number of distinct colors that can be represented by a pixel depends on the number of bits per pixel (bpp). The maximum number of colors a pixel can take can be found by taking two to the power of the colour depth. For example, common values are:

8 bpp, $2^8 = 256$ colors
 16 bpp, $2^{16} = 65536$ colors, known as High colour or Thousands
 24 bpp, $2^{24} = 16,777,216$ colors; known as True colour or Millions
 48 bpp; for all practical purposes a continuous color space; used in many flatbed scanners and for professional work

Let us represent a wireless node by 2 pixels in a video frame as in Fig. 51. The pixel coordinate (in the frame) is determined during provisioning the node in the Database (say HLR or HSS) [10,11,12].

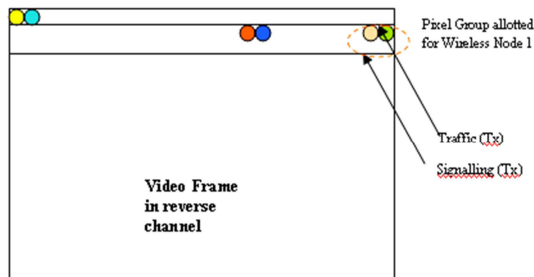


Figure 51 : Video Frame in Reverse Channel

The video signal carries the picture and colour information for each and individual pixels. Each wireless node in the colour space domain is identified by a specific colour level (a combination of hue and saturation level) and allotted a particular pixel group (which includes 2 pixels as shown in the figure) belonging to a specific coordinate in the colour space domain [3]. A specific colour level (band) is assigned for signaling information and an adjacent band for traffic information (to be carried in the adjacent pixel). The wireless node can clamp on to a specific pixel group occupying a specific coordinate of the video frame. The number of simultaneous users that a video frame can hold is one half of the resolution supported by the video frame (since each wireless node uses 2 pixels for communication). In case of a video frame in the forward direction, the wireless nodes generates the

pixel information for signaling pertaining to the type of operation, say for call generation, powering up. After the signaling information is exchanged with the network (and the end device) and the traffic/data channel is realized, the pixel meant for traffic channel (in Tx and Rx for forward and reverse frame) carries traffic / data information. In case of the video frame in the reverse direction, a particular wireless node scans the pixels available in the video frame (the pixel clock inside enables sampling at the same frequency as the source), picks up whether it obtains any pertaining signaling information (according to the hue and saturation level allotted to itself) in the given pixel (meant for carrying the signaling information) and then responds to it. Subsequently the traffic channel is seized and the one pixel meant for carrying the traffic information in the reverse direction starts carrying the traffic information (say for ring signal) and the wireless node accepts and processes that accordingly. The model, as in Fig. 52, explains how the video frames are synthesized for mobile / computer nodes (eg provided for wireless communication).

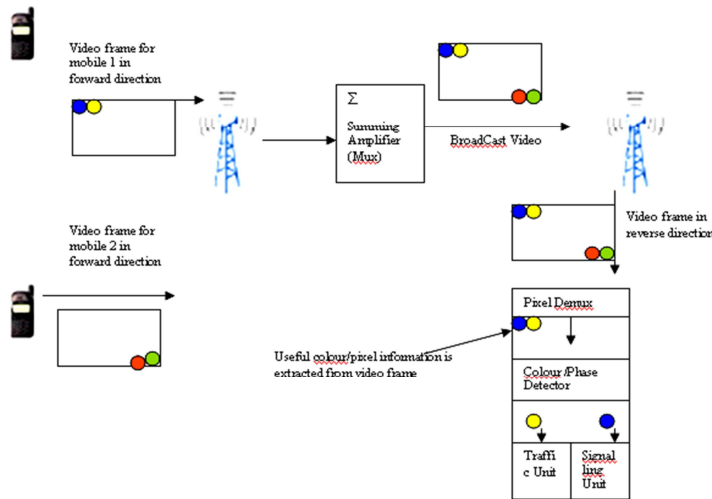


Figure 52: Synthesis of Video Frames in the Forward and Reverse direction for Mobile Nodes

Apart from assigning the pixel coordinates during provisioning a mobile node in a network, a specific colour band is also allotted for the node (as indicated previously), which is used as its identity in the network. A band of saturation level is spread across the colour vector, a sub-band of which is meant for signaling and the remaining for bearer traffic. Hence for the mobile node, the MIN/MSI/MSIDN needs to be a function of the phase of the colour vector and the band of saturation level for transcoding voice and signaling.

$$\text{MIN (E.212 NP) / MSIDN (E.164 NP)} = f(\emptyset, S_b)$$

Where $S_b \Rightarrow$ the band of saturation level allotted to the user for signaling and bearer data

The coordinate of the mobile station in the colour circle are colour level ranges (bands) allocated for each user, as depicted in Fig. 53.

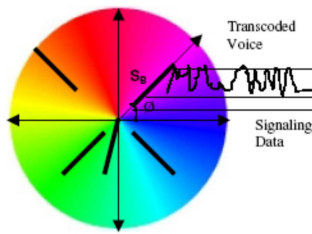


Figure 53: The users defined in the colour circle

5.4. Architecture of a WLL network based on Colour Signals

This section discusses the design philosophy of a WLL network applying the technique of colour pixel multiplexing.

As this is based on the existing video technology, our aim is to propose an architecture which is based on the existing cable television network. The proposal is to reuse the backhaul available for the Television / cable networks and to have the last mile radio access based on Colour Pixel Multiple Access Technology designed for WLL access.

The diagram below in Fig. 54 shows a network of TV Japan, and how it caters to three different categories of television subscribers over three different physical channel, Cable / Internet and satellite. With the proposed multiple access scheme, it will be possible to use the existing television network for implementation in voice / data communication in a typical WLL implementation.

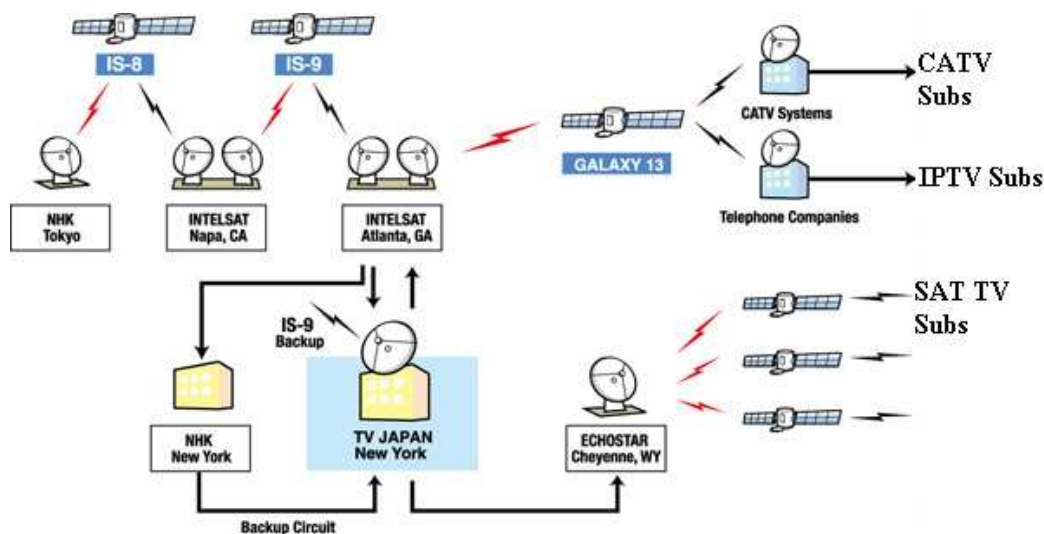


Figure 54: A typical Sat TV network

At a high level, the implementation below in Fig. 55, indicates the usage of the Digital Television Network for the backhaul for the proposed WLL Network. The WLL operator will need to subscribe for a Television channel which acts as a carrier for voice and data. The operation should be low range

(within a mile) and Near Line of site (NLOS) operations. The aim is to avoid the channel impairments due to the physical effect which magnifies with distance and obstructions.

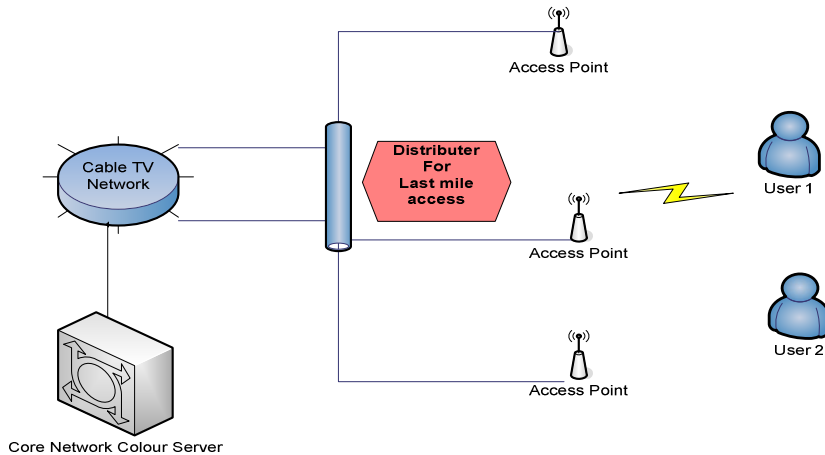


Figure 55 : Cable TV network for WLL Transport

For the access network, we have the BTS termed as Video – BTS (V-BTS), the only difference with GSM/UMTS/CDMA being the fact that the network area is not divided into orthogonal cells and there is no BSC in place. The V-BTS placed all over the network coverage area can establish an uplink and downlink channel in UHF / VHF with the handsets. As the BTSs are not serving any cell, they need not have any specific BTS-ID for addressing. The BTS does not transmit the BTS-ID and other essential parameters (SID / NID / MCC/MNC) in the SPM /ESPM messages (CDMA 2000), and does not exchange any specific BTS/Location related information with the mobile station. The MS is agnostic to its location in the coverage area, as it is not served by a particular cell. The V—BTS s are dumb video repeaters placed all across the network area. All the repeaters placed across different zones transmit the same chrominance signals which carry the voice / data intelligence for the wireless users. The number of users catered by a single colour circle pertaining to the allocated bandwidth of 37 Mhz depends upon the colour depth supported by the video technology and the resolution supported.

The number of distinct colors that can be represented by a pixel depends on the number of bits per pixel (bpp). The maximum number of colors a pixel can take can be found by taking two to the power of the colour depth. Colour depth is a computer graphics term describing the number of bits used to represent the colour of a single pixel in a bitmapped image or video frame buffer. This concept is also known as bits per pixel (bpp), particularly when specified along with the number of bits used. Higher colour depth gives a broader range of distinct colors.

The architecture of the WLL at a high layer showing the core network and the radio network components is shown in Fig 56. It utilizes Colour Pixel Multiple Access (CPMA) for last mile radio access. Please note that this diagram focuses more on the logical interfaces between the networks, but not on the Back Haul (TV network).

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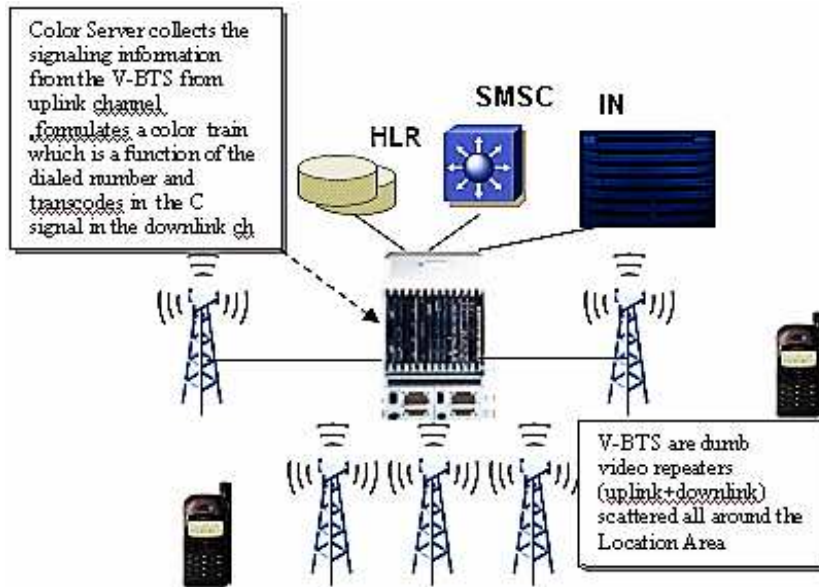


Figure 56 : The WLL network

Each handset after receiving the chrominance signal demultiplexes and extracts the information from the pixel allotted to itself and then processes the colour signal (in terms of R – G – B Levels) pertaining to the specific hue and the band of the saturation level in digitized form. Within the specific band of the saturation level allotted for the subscriber , say Band A (ranges 10% to 35 %) , a specific sublevel say Band A1 carries the signaling information (for paging , Alert with Info , etc) and the Band A2 , carries the digitized voice information. The heart of the network is a colour server which processes all the colour information. The colour server interoperates with the Mobile Station through the V-BTS / V-BSC and exchanges the SS7 signaling information with the core network accordingly. The speech circuit between the A and the B party is established on the band of saturation level tied up with the colour level that has been assigned for both the parties.

The primary functions of the colour server are the following [3].

1. *Performs colour signal processing for the access network.*
2. *Formulates the colour train which is a function of the dialed E.164 mobile number (of the same serving market).*
3. *Performs the signaling operations (on SS7) with the core network.*
4. *Interoperates with the V-BTS/V-BSC for the air interface related operations.*
5. *Interfaces with the Legacy Networks (on associated mode of signaling and PCM Voice trunks on F Links and on A links on quasi associated mode).*

6. *Call Data Record generation for Mediation / Billing.*
7. *Call routing for the legacy network.*
8. *SSP Functionality for Intelligent network operations.*
9. *Supports Supplementary Services*

The flow showing all the interaction between the Mobile Station and the network for call maturation is depicted in Fig 57.

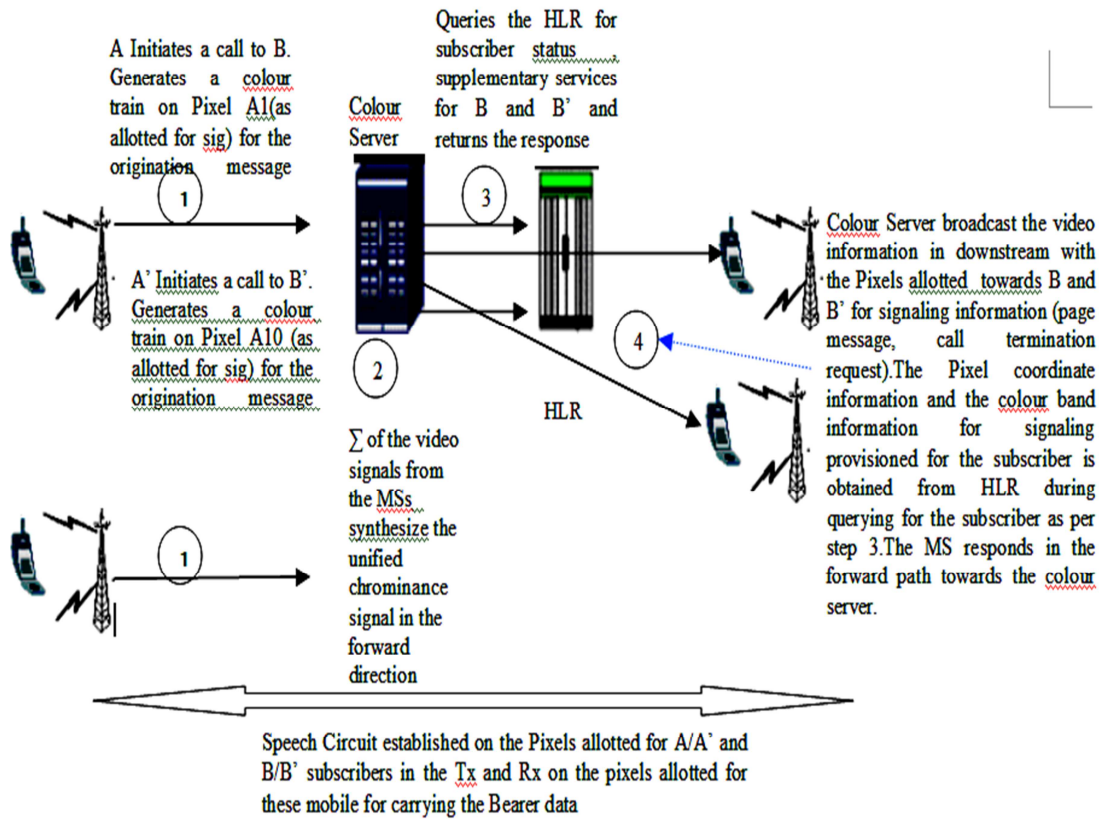


Figure 57 : The call flow showing the interaction between Mobile Station and the Network

The following diagram in Fig 58 shows the internal constituents of the Mobile Station and the Call Processor of the Colour server.

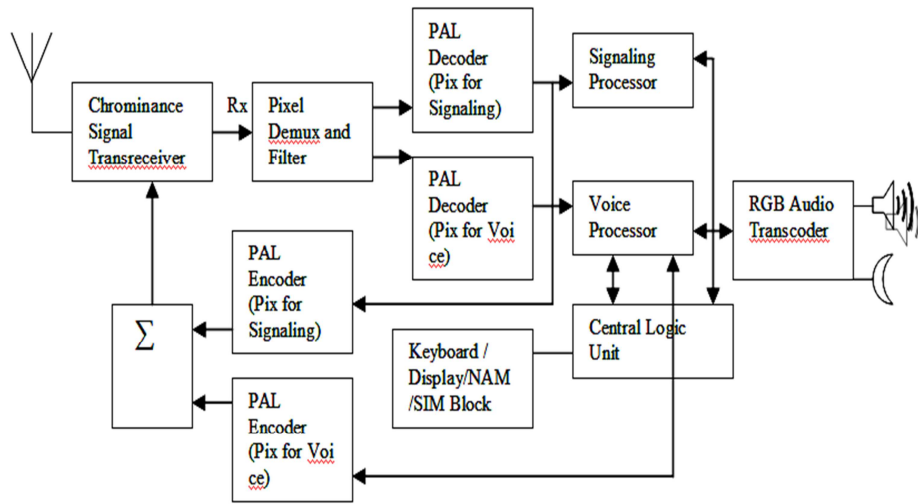


Figure 58 : Internal blocks of a mobile station

5.5. System Model

A subscriber is assigned a time slot (i.e., pixel) w.r.t a particular horizontal line scan. So the horizontal line number and the pixel coordinate of time slot are fixed for the customer during provisioning in the network. Chrominance Signal C (composite video) is continuous TDM signal. The Time slots are calculated in terms of resolution, i.e. the number of TS / Pixels, supported per line and the number of lines. For a given time slot for a particular subscriber, i.e. pixel in a video frame, the following are the variables, as in Fig 9.

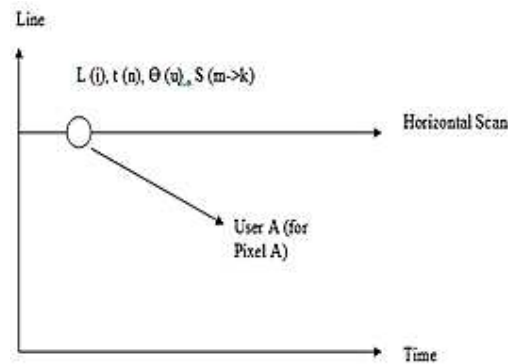


Fig 9. Depiction of a video frame in the X-Y Coordinates

- $L(i)$ represents Line i
- $t(n)$ represents the time instant (n) when the color information is brought for the particular pixel.
- $\Theta(s)$ depicts the hue assigned for the subscriber
- $S(m \rightarrow k)$ represents the band of saturation level allocated for the subscriber. In our initial discussion we will keep m as 0 and k as 100 % to keep the things simple. In actuality,

multiple subscribers will be accommodated over the same hue and different bands of saturation level with some amount of dead spaces between each individual band of saturation level.

Let C be the instantaneous signal for Pixel A at $t(n)$.

The next instantaneous C signal will arrive for the given pixel after the horizontal scanning is completed for all the other pixels in the video frame.

Let R be the resolution of the Frame = $L \times p$

(As mentioned above, L represents the line number and p represents the number of pixels per line)

In that case, if R is the resolution of the frame (say 640 pixels per line and 485 lines), the time instant at which the next information arrives for the given Pixel $A = t(n - (R-1))$.

Hence the C signal w.r.t a Particular user A (pertaining to Pixel A)

$$C(A) = C_1 t(n) + C_2 t(n - (R - 1)) + C_3 t(n - 2(R - 1)) \dots C_k t(n - k(R - 1)) \quad [10]$$

Hence the C signal for all the users, which is half of the resolution of the video frame (as one pixel will be allocated for signaling and the adjacent one for voice / data)

$$C = \sum C_1 t(n) + C_2 t(n - (R - 1)) + C_3 t(n - 2(R - 1)) \dots C_k t(n - k(R - 1)) \quad [11]$$

where $A = 1t$

Now let us analyse the contents of C signal.

C signal = Chrominance signal + Y signal + Horizontal and Vertical & Blanking Pulses which makes it a composite video signal.

$$\text{So let } C = C_r + C_y + H_p + H_v \quad [12]$$

(H_p and H_v being the horizontal sync and the vertical pulse information)

Specifically with respect to an individual user, the H_p and H_v are not to be considered.

These two parameters will be considered when the entire composite signal is expressed.

Cr signal:

Let us denote with $s(t)$ the audio signal to be transmitted. The user i is assigned a fixed Hue H_i (0-360°) and the entire saturation band (0-100%), which is modulated by the information signal $s(t)$. The saturation level $S_i(t)$ and the hue are related to the component UV by the following relationships:

$$U = S_i(t) \cos H_i$$

$$V = S_i(t) \sin H_i$$

$$Y = 0.299 * R + 0.587 * G + 0.114 * B$$

$$U = 0.436(B - Y)/(1 - 0.114)$$

$$V = (R - Y)/(1 - 0.299)$$

And that R, G, B , range between 0 and 1, then:

Therefore, $S_i(t)$ ranges between $[0, 0.753]$ and $s(t)$ must be normalized to fall into that range. So, if V is the maximum value of $s(t)$, then the chrominance component U V can be written as a function of the information signal as follows:

$$U = s(t) * (0.753/V) * \cos H_i$$

$$V = s(t) * (0.753/V) * \sin H_i$$

In the NTSC system, the U V components are combined into a single modulated Chroma signal by using quadrature modulation onto a continuous wave color subcarrier. The modulated Chroma signal can be written as:

$$Cr(t) = s(t) * (0.753/V) * \cos H_i \sin(\omega t) + s(t) * (0.753/V) * \sin H_i \cos(\omega t)$$

Hence Cr signal is primarily a function of the color difference signals $R-Y$ and $B-Y$. Green signal is not transmitted but synthesized from R , G and Y information through a proper expression, Cy or Y

$$Y = 0.299 * R + 0.587 * G + 0.114 * B$$

Now,

$$R = 1.164(Y - 16) + 1.596(V - 128)$$

$$= 1.164(Y - 16) + 1.596(s(t) * ((0.753/V) * \sin H_i - 128)) \quad [13]$$

$$G = 1.164(Y - 16) - 0.813(V - 128) - 0.391(U - 128)$$

$$= 1.164(Y - 16) - 0.813(s(t) * ((0.753/V) * \sin H_i - 128)) - 0.391(s(t) * (0.753/V) * \cos H_i - 128) \quad [14]$$

$$B = 1.164(Y - 16) + 2.018(U - 128)$$

$$= 1.164(Y - 16) + 2.018(s(t) * ((0.753/V) * \cos H_i - 128)) \quad [15]$$

Hence,

$$Y = 1.164(Y - 16) + 0.299 * 1.596(s(t) * ((0.753/V) * \sin H_i - 128)) + 0.59 * 0.813(s(t) * ((0.753/V) * \sin H_i - 128)) - 0.59 * 0.391(s(t) * ((0.753/V) * \cos H_i - 128)) + 0.114 * 2.018(s(t) * ((0.753/V) * \cos H_i - 128))$$

Hence Y = Expression K (by resolving the above equation).

Hence the final equation for the composite video signal for the different users allocated pixels / time slots in the video reference frame is

$$C = \sum [Cr + Cy]_1 t(n) + [Cr + Cy]_2 t(n - (R - 1)) + [Cr + Cy]_3 t(n - 2(R - 1)) \dots \dots \dots [Cr + Cy]_k t(n - k(R - 1)) + H_p \times L + V, \quad [16]$$

where $A = 1$ to $R/2$

A depicts the number of users served by one video frame.

5.6. Example

Let us take the example of a SXGA video signal, which can be transmitted on wireless or CAT 5 cable.

The parameter values / characteristics are as follows in Table 3

Parameter	Value
Active Horizontal Pixels	1280
Active Vertical Pixels	1311
Total Horizontal Pixels	1720
Total Vertical Pixels (VLT)	1067
Frame Rate (FR) (Hz)	76
Horizontal Rate (KHz)	81.1
Pixel Rate (Mpixels/second)	139.5
Signal Bandwidth (BWS) (MHz)	51.9

Table 3 : Parameter values for the example to derive the number of users that can simultaneously use the service

Total Number of Pixels in each frame = 1280×1311

We emphasize on two factors

1. The number of users that can talk simultaneously within a video frame

- 1 One subscriber requires data @ 8 KHz (same as what voip uses at the moment).
- 2 The Pixel Pixel rate – The rate at which the pixels arrive in the display device = 139.5 Mhz for $1280 \times 1311 @ 76\text{Hz}$
- 3 So the number of subscribers that can theoretically talk simultaneously with one colour circle= 17438

Say if 15 percent of the bandwidth is consumed for signaling, then the total number of users that can talk simultaneously = 14822 Calls per second.

2. The number of subscribers that can be provisioned in a video frame.

V freq refresh rate = 76 Hz

H Freq rate = 95 KHz which translates to 1280×1024 resolution

Our point of interest is H Freq rate

We can employ multiple pixels, say along the same column and equally spaced in the video frame.

We need around 8 KHz bandwidth for a subscriber.

Horizontal refresh rate = 81.1 KHz.

Hence if we assign around 10 pixels per user, which are equally spaced in a column, the bandwidth of around 8 KHz or more will be achieved .

Hence if the resolution of the video frame is 1280x1311, the number of users that can be accommodated / provisioned per colour circle = $1310720 / 10 = 167808$

5.7. Signal Bandwidth

The signal bandwidth comes from the following equation,

$$BWS = 1/2 [(K*AR*(VLT)^2*FR)*(KH / KV)] = 51.9MHz \quad [17]$$

Where BWS = Signal bandwidth K = Kell factor, Visual information is lost due to the probability that some of the video information will be displayed during the retrace rather than the active portion of the scan line. Assuming 30 percent of the visual information is loss, we have K = 0.7.

AR = Aspect ratio (the display width divided by display height) = 1.33 VLT = Total number of vertical pixels = 1067 FR = Frame rate or refresh rate = 76 KH = Ratio of total horizontal pixels to active pixels = $1720/1280=1.34$ KV = Ratio of total vertical lines to active lines = 1.04

The 1/2 factor comes from the highest frequency component in a video signal occurring when alternating black and white vertical lines with a width of one pixel are displayed on the screen. Because it takes two lines to form a complete cycle, the highest frequency is one-half the pixel rate.

The Kell factor represents the effect of reduced visual resolution primarily due to the line-scanning structures. Visual information is lost due to the probability that some of the video information will be displayed during the retrace instead of the active portion of the scan line. Even though it may seem like half the information would be lost because there are equal number of scan and retrace lines, empirically it has been shown that about 30% is lost to this effect, yielding a Kell factor of about 0.7.

The frame rate is the rate at which each complete set of scan lines is displayed. Because a set of scan lines makes a complete picture, this can be thought of as the picture-update rate. Most television signals are in an interlaced format. This is where each picture (or frame) is divided into two fields, each with half of the vertical scan lines. This doesn't affect the calculation as long as the actual frame rate is used in the equation. Let us recall that the frame rate is equal to half the field rate.

In our research work, we will however prefer progressive scanning and not interlacing, mainly because of its high frame rate (60 frames per second) and the complexity of multiplexing / demultiplexing the 1/2 frame lines.

5.8. Amplifier bandwidth and slew rate requirements

To maintain video signal integrity, we need to maintain 0.1dB bandwidth to the signal bandwidth (BWS). When selecting amplifiers, special attention should be given to its frequency response characteristics; for instance, for a signal pole amplifier the 3dB bandwidth required to handle 51.9MHz is $6.5*51.9MHz = 337MHz$. For multiple pole amplifiers (most modern high speed amplifiers are multiple pole amplifiers), the 3dB bandwidth should be set to 3 times the signal bandwidth, which for the previous example would be 155.7 MHz. The slew-rate can be calculated from the signal amplitude and pixel rate. So to maintain video signal integrity with a pixel rate of

139.5MHz while allowing the signal to complete its transition during $\frac{1}{4}$ of a clock period use the following equation.

$$\begin{aligned} \text{Slew Rate} &= 1/(\frac{1}{4} * \text{Pixel Time}) \\ &= 1/(\frac{1}{4} * (1/139.5\text{MHz})) = 558\text{V}/\mu\text{S} \end{aligned} \quad [18]$$

5.9. Conversion of voice to colour signals

When a subscriber is provisioned in a network, and in the EEPROM /SIM card of the WLL handset, he/she is assigned the following parameters.

1. International Mobile Subscriber Identity
2. Pixel Coordinate: This will identify the two pixels, one for voice/user data and one for signalling in the video frame for the forward and the reverse channel. The voice/ data and the signalling information for the particular subscriber will only be conveyed within the pixels allocated for the subscriber.
3. Colour: The band of colour (a particular hue and a band of saturation level) is allocated for the subscriber. Colour is a combination of hue (the tinge) and saturation level (how much white is mixed with the colour. The hue is depicted by the phase angle of the colour vector in the colour circle. The saturation level is depicted by the amplitude of the colour vector. The voice/data is carried by Amplitude modulation (or by modulation to two fixed levels, 0 and 1 in case of digital data) within the allocated band of saturation level pertaining to the hue/tinge.

The video (chrominance) signal is a continuous signal consisting of video frames refreshed at a specific rate.

$$C = \sum v(t), t = 0 \text{ to } n$$

Both in the forward and the reverse channel, all the subscribers use the same channel frequency. Because the pixel coordinates are unique and fixed for the subscribers, the colour information of the pixels are summed up in the forward and the reverse direction by a summing amplifier/ video mixer to form an entire video frame carrying the information for all the users.

It is like two video channels having the same frequency and you see a juxtaposed image belonging to the two video channels.

Hence the variables for a given subscriber are

1. Pixel coordinate, say (x_i, y_i) . For signalling and voice /data, two pixels are utilised, hence the pixel coordinate for voice /data = (x_i, y_i) and signalling = (x_j, y_j)
2. Θ_s , ie the hue allocated for a user (there can be multiple users allocated the same hue but different frequency bands).
3. Band of Saturation level for carrying the signalling /voice information (A_s).

The colour processor needs to take care of all these variables for all its users when it requires formulating a video train for data/voice transmission in the forward or reversing direction.

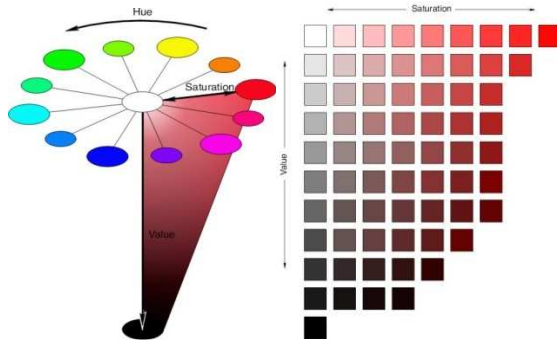


Figure 59: The Hue and Saturation level relationship in colour space domain

The above diagram in Fig 60 shows the hue and saturation level relationship w.r.t to the colour space domain. The saturation level is measured from 0 to 100. Saturation level had to start from a finite value, and not 0 or null value for a particular subscriber, as 0 implies pure white with no hue information. Hence we cannot differentiate the subscriber for that purpose.

Say the Saturation level starts from 1 and goes, and the colour vector takes a particular phase pertaining to the hue, as defined for the subscriber.

Voice can be carried as Analog voice signal on AM or might be digitized , which will necessitate only two discrete saturation levels for carrying the voice.

5.10. Amplitude Modulation (based on modulation of the saturation level) on the chroma signal for voice / data traffic conveyance

For Amplitude Modulation , the carrier for the voice / data signal for a specific user is again the chrominance signal of a particular phase angle (operating within the domain of the band of saturation level allocated for the user) . The carrier frequency will be 3.34 Mhz , that of the Chroma signal.

The Amplitude modulation can be used here is a Double side band AM or a Reduced Carrier Transmission AM.

A carrier wave is modelled as a simple sine wave, such as:

$$c(t) = C \sin(\omega_c t + \phi_c) \quad [19]$$

where $\omega_c = 2\pi f_c$

where C and ϕ are arbitrary constants which represents the carrier amplitude and initial phase.

Let $m(t)$ represent a simple audio waveform that needs to be transmitted , and let M represent the largest magnitude

$$m(t) = M \cos(\omega_m t + \phi_m) \quad [20]$$

It is assumed that $\omega_c \gg \omega_m$, as audible frequency ranges from 20 Hz to 20,000 Hz and the carrier frequency of the Chroma signal is in the order of Mhz.

The Amplitude modulation is hence provide by the equation

$$y(t) = A + m(t). c(t), \quad [21]$$

where A represents a constant

Therefore, the modulated signal has three components, a carrier wave and two sinusoidal waves (known as sidebands) whose frequencies are slightly above and below.

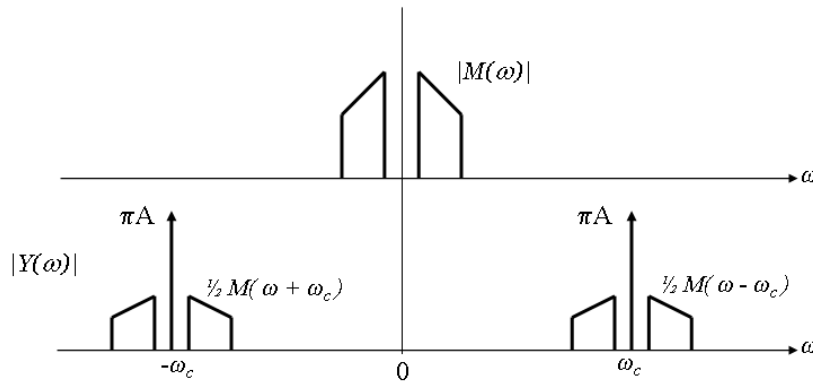


Figure 60: 2 sided spectrum of AM signal

Fig. 61 demonstrates a (2-sided) spectrum of an AM signal. It has two groups of components: one at positive frequencies (centered on $+\omega_c$) and one at negative frequencies (centered on $-\omega_c$). Each group contains the two sidebands and a narrow component in between that represents the energy at the carrier frequency. Therefore, we see that an AM signal's spectrum consists basically of its original (2-sided) spectrum shifted up to the carrier frequency.

5.11. Spectral Efficiency

Spectral Efficiency for CPMA

Spectral Efficiency = (Channel Throughput/Channel Bandwidth)

[22]

= $14822 \times 8 \text{ kHz} / 51.9 \text{ Mhz} \times 1$ (14822 parallel users as in the previous calculation , 8 Khz voice band per user & freq reuse factor of 1)

= 2.28 b/s/hz/colour circle

The pixel rate for the present day video technology was however determined to eliminate flicker . But now as we are employing the technology for a different purpose (ie, as a carrier and addressing mechanism for a wireless network) we can actually achieve a much higher data rate .

Besides the spectral efficiency, there are other factors which need to be considered where we will benefit from this technology.

5.12. Advantages of CPMA in light of the various factors essential for the Network Operations

- **Frequency Planning:** A substantial amount of the effort in cellular systems is devoted to managing interference through the use of a “reuse pattern.” Traffic channels are partitioned into groups

where, say, each group has one-seventh of the total radio resources, and channel groups are assigned to base stations in the network in such a way that any two cells using the same channel group are not adjacent. The resulting spatial separation ensures that the energy being used for a conversation in one cell has been sufficiently attenuated by the time it reaches another cell using the same channel that it does not pose significant interference. Reuse provides interference management, but at the expense of operational complexity and base station capacity. A given base station's channel group is a small fraction of the total system resources.

- With the CPMA, frequency planning becomes easier. The whole bandwidth is utilised by all. Moreover the need for cell planning and frequency reuse is lessened. Hence the factors like frequency reuse, frequency hopping, co-channel interference etc become less important, which implied that we achieve a better spectral efficiency and a better SNR.
- Power control: Power control is a technique whereby the transmit power of a base station or handset is decreased to near the lowest allowable level that permits communication. This reduces interference levels in the network, increasing spectral efficiency. Power control is used in both GSM and CDMA systems on both the uplink and downlink. CDMA systems require particularly fast and precise power control – many users share the same RF spectrum, and the system capacity is thus highly sensitive to inadequate interference control. The base station in case of the CPMA ,does not need any discrete address .All the base station belonging to one colour circle will have the same identification/address .They are in fact video repeaters spread across the colour circle.Keeping in mind the reflection factor, the video repeaters can be placed as close to the MS. As the signaling requirement is much less in this case, hence power is conserved much better.
- Sectorization: Sectorized antenna systems take a traditional cell area and subdivide it into “sectors,” each covered by its own directional antenna sited at the base station location. Operationally, each sector is treated as an independent cell. Directional antennas have higher gain than omni-directional antennas, all other things being equal. Hence the range of these sectors is generally greater than that obtained with an omni-directional antenna, roughly 35 percent greater. Sectorized cells can increase spectral efficiency by reducing the interference presented by the base station and its users to the rest of the network, and they are widely used for this purpose. Most systems in commercial service today employ three sectors per site. Although larger numbers of sectors are possible, the number of antennas and quantities of base station equipment become prohibitively expensive for most cell sites. Cell sectorisation is not required in our case as we work with the entire spectrum.

7

Multisource Voice coding technology using colour coding technique for high definition voice transmission

6.0. Introduction

This chapter addresses a mechanism for joint source channel coding for transmission of audio signals transcoded to lossless video signals[4]. The audio information is mapped into colour information with the help of the intelligence derived from a phase vocoder. Each pixel in the video frame, meant for the transmission of the audio signals over video, pertains to a specific frequency in the time domain. A couple of harmonic frequencies along with the fundamental frequency is required to be processed for carrying the voice intelligence. Hence, for conveying a speech signal for a given user, a number of colour pixels are utilised. This implies that the number of users that can be accommodated depends on the resolution that is supported by the video encoding scheme and the QOS needed for carrying the voice. At the source, we actuate phase vocoding and colour pixel multiplexing. This part can be termed as channel coding as we encode each harmonic in a suitable pixel and thereby render redundancy to the original information. The video stream which comprises number of pixels per frame pertaining to the resolution is multiplexed at the source and then transmitted using an appropriate lossless source coding scheme. In this chapter, we consider MPEG 4 AVC which can attain nearly 3:1 lossless compression with 4:4:4 encoding of YCbCr signal.

6.1. Description

The problem of joint source channel coding had been an interest for the scientists for decades. A joint source channel coding mechanism for voice transmission on colour signals is discussed in this chapter [4]. At the channel coding level, we use a phase vocoder which converts the voice frequencies to equivalent phase time domain model. The encoding is done with the help of STFT (Short Time Fourier Transform). The phasors derived after the STFT are dynamically plotted to a colour circle. Each phasor information, ie the phase and the amplitude (in the time domain), is expressed in terms of colour and is transmitted over a pixel. The number of pixels contained in the video frame depends on the supported resolution. We actuate pixel multiplexing at the audio source, whereby the audio frequencies of multiple sources are multiplexed at the pixel level to formulate a video frame. The dynamic audio information from multiple sources are ported to video. At the source coding level, we propose a lossless video transmission using MPEG_4 AVC compression mechanism by which we can attain nearly 3:1 compression. At the receiver side, we first actuate MPEG4 decompression and

decoding. Then the time phase model for the phase voice decoder is reconstructed from the colour signal. Typically the tinge of the colour depicts an instantaneous phase and the saturation level brings the modulo of the signals, ie the amplitude. For a single voice signal, each phasor depicts a particular frequency, which is carried by a particular pixel. The number of Harmonic frequencies that can be carried along with the fundamental frequency (f_0) for a voice user depends on the type of application or the QOS settings for the voice user. Each frequency converted to the phase time domain can be carried by a single pixel. So as the number of harmonic frequency increases, the number of pixels required to be allocated for a given user increases. Each pixel corresponds to a frequency bin of the phase vocoder. The decoding happens with Inverse Short Time Fourier Transform (ISTFT) to derive the equivalent frequencies in the time domain.

6.2. The channel coding scheme : Vocoding and conversion of speech to colour signals

As sound (in the physical sense) is a wave, it has many properties that are analogous to the wave properties of light. Think of a prism, which bends each color through a different angle and so decomposes sunlight into a family of colored beams. Each beam contains a “pure color,” a wave of a single frequency, amplitude, and phase. Similarly, complex sound waves can be decomposed into a family of simple sine waves, each of which is characterized by its frequency, amplitude, and phase. These are called the *partials*, or the *overtones* of the sound, and the collection of all the partials is called the *spectrum*. This prism effect for sound waves is achieved using the Fourier transform.

Mathematically, the Fourier transform of a function is defined as

$$\begin{aligned} X(f) &= \int_{-\infty}^{\infty} x(t)e^{-j2\pi ft} dt \\ &= \langle x(t), e^{j2\pi ft} \rangle \end{aligned} \quad [23]$$

The short-time Fourier transform (STFT) is often used when a signal is too long to be analyzed with a single transform or when it is desirable to have better time localization. The idea is to use a window function $w(t)$ that zeroes all but a short time interval. All events in the FFT are then localized to that interval. The windows are shaped so that when they are overlapped (shifted by S samples and summed) their sum $\sum_n w(t-Ns)$ is constant for all t . This is shown in the following figure (Fig. 62) where the windows are overlapped by their half phases.

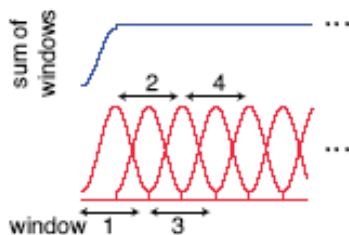


Figure 61 : Overlapped windows in FFT.

A short-time Fourier transform (STFT) is performed on a windowed time-domain real signal to obtain a succession of overlapped spectral frames (synthesis stage). The time delay at which every spectral frame is picked up from the signal is called the *hop size*. The time domain signal may be rebuilt by

performing an inverse Fast- Fourier transform on all frames followed by a successive accumulation of all frames (an operation termed *overlap-add*) (resynthesis stage).The process is shown in Figure 63.

Knowing the modulus of every bin is not enough: the phase information is necessary for a perfect recovery of a signal without modification. Furthermore the phase information allows an evaluation of 'instantaneous frequencies' by the measure of phases between two frames. Thus, in a traditional phase-vocoder implementation the output of the analysis should be in explicit polar form (moduli and phases) in order to achieve fine-grain tracking of frame by frame frequency changes.

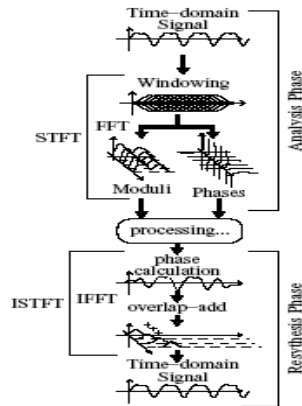


Figure 62: Short term Fourier Transform and the Inverse

The transform of a complex sinusoid to the equivalent moduli and phase is shown below sampled through n number of windows.

complex sinusoid: $x(n) = Ae^{j\omega_x n}$, A : amplitude, ω_x : radian frequency

window: $w(n)$, $n = 0, 1, \dots, M-1$

$$\begin{aligned} X_w(\omega) &= \sum_{n=-\infty}^{\infty} x(n)w(n)e^{-j\omega n} \\ &= A \sum_{n=-M/2}^{(M/2)-1} w(n)e^{-j(\omega-\omega_x)n} \\ &= AW(\omega-\omega_x) \end{aligned}$$

[26]

6.3. Deriving the Amplitude and Phase from the FFT Bins to derive the color signals

A function converts each FFT bin from rectangular to polar form to get the magnitude and phase in each FFT bin .

The instantaneous frequency (for the kth bin) is derived latter on

$$A_k(m) \triangleq |\tilde{x}'_m(e^{j\omega_k})| \quad [25]$$

$$\Theta_k(m) \triangleq \angle \tilde{x}'_m(e^{j\omega_k}) \quad (\text{radians}) \quad [26]$$

$$F_k(m) \triangleq \frac{\Theta_k(m) - \Theta_k(m-1)}{2\pi RT} \quad (\text{Hz}) \quad [27]$$

$M \Rightarrow$ Number of samples

$R \Rightarrow$ Down sampling factor

Where

$$\omega_k = 2\pi k f_s / N$$

And

$$f_s = 1/T$$

is the sampling rate in Hz. The STFT bin number is k

The instantaneous color vector for a single sample for the k th bin is $C_k = f(\theta_k, A_k)$.

For t^{th} time, there are $1, 2, \dots, k$ STFT bins for a single users. The bins carry fundamental frequency (f_0) plus all the other harmonics / semitones. The number of allowed semitones along with the fundamental frequency will depend on the QOS configuration for the users. Accordingly, the number of pixels (in the t^{th} video frame) required by a given user can be determined. Each pixel can carry information for a single STFT bin for t^{th} time. Fig 64 shows how the colour pertaining to each STFT bin forms the array holding the colour information of the entire video frame. The Video frame holds the dynamic audio information.

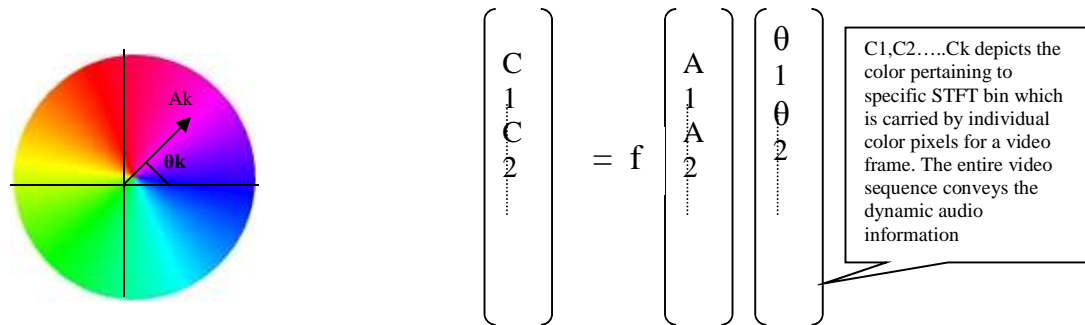


Figure 63: The colour information in the STFT bin.

Figure 65 shows the full scheme where the audio source is split up in multiple sub harmonics by performing Short Term Fourier Transform. Each Harmonic information derived as a function of

Amplitude and Phase is ported to pixels to form dynamic colour information as video frames. This is described in more details as follows.

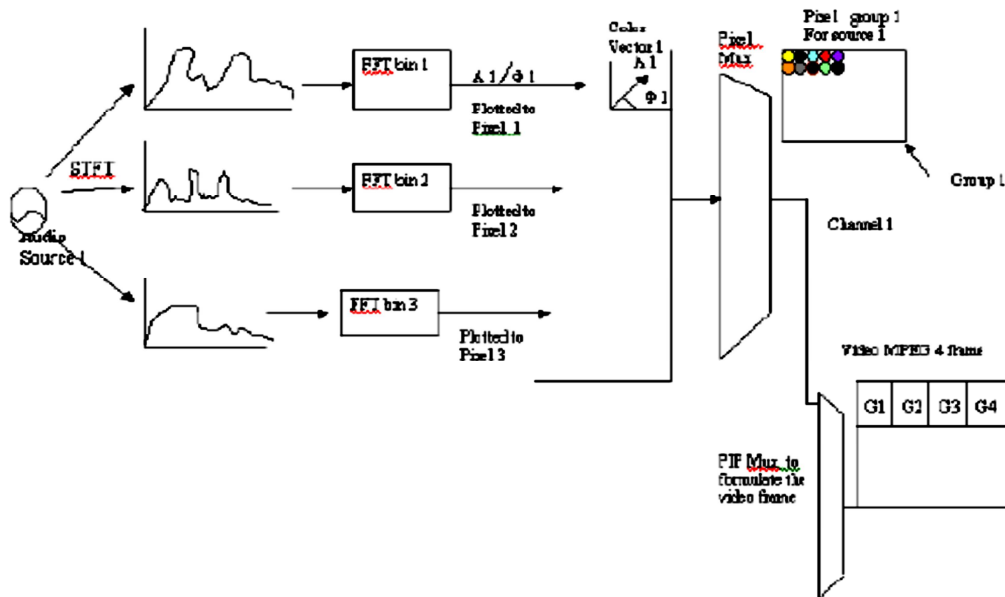


Figure 64: The complete scheme of STFT and ISTFT of colour pixels

Audio signals from different sources are sampled with the help of sampling windows and split up in terms of different phase and moduli of the different frequencies/harmonics/semitones. Each of them (for a given user) corresponds to a specific FFT bin (out of a FFT bin bank) meant for the specific frequency.

The output of the FFT bin, in terms of the phase angle and the amplitude corresponds to a specific colour. The tinge of the colour is dependent on the phase angle and the amplitude determines the saturation level, ie, how much white is mixed with the colour level. This colour output for a FFT bin is plotted against a pixel.

The digitized output colour levels of the FFT bin bank for a specific user is plotted against a group of pixels, with the aid of pixel multiplexing. Accordingly, the Y,Cb,Cr signals for the source is formed.

The mapping of the FFT bin to the coordinate of the pixel is done by the array function. Each user is allotted a channel frequency. The output of different users (colour plotted in a pixel group) is fed to an array of PIP (Picture in Picture) modules. The PIP modules belonging to the different channel frequencies (for different sources) jointly make a video frame. The video frames in the time domain contrsue a streaming video which is encoded in MPEG-4 AVC.

The reverse process is followed at the receiving end to derive the audio signals for different receivers in the order shown below.

1. MPEG4 decoding

2. PIP to targets mapping
3. Pixel demultiplexing and routing to the Inverse STFT bins.
4. Inverse STFT
5. Construct the original audio signal for the different users.

The source coding scheme - MPEG-4 AVC Lossless video coding

The basic steps of MPEG 4 AVC lossless coding [14] involves mode selection, temporal compression, spatial compression and entropy encoding. As in Fig. 66, the first process, ie, mode selection divides the frame in subframes with a pixel matrix of 16x 16, which are called macroblocks. The encoder then determines whether the encoder will go for intra frame or inter-frame encoding for the macro block.

If Interframe encoding is chosen, then motion compensation techniques are used for removing temporal redundancy and discrete cosine transform followed by quantisation for removing spatial redundancy of the video images.

If Intra frame encoding is chosen, the information contained in the current video frame and produces a compressed result called I-Frame. Intra frame encoding can use only the information in as many as five other frames , occurring before or after the current frame. Compressed results that use the data from previous frames are called P_Frames while that uses the data both from before and after the current frames are called B-Frames.

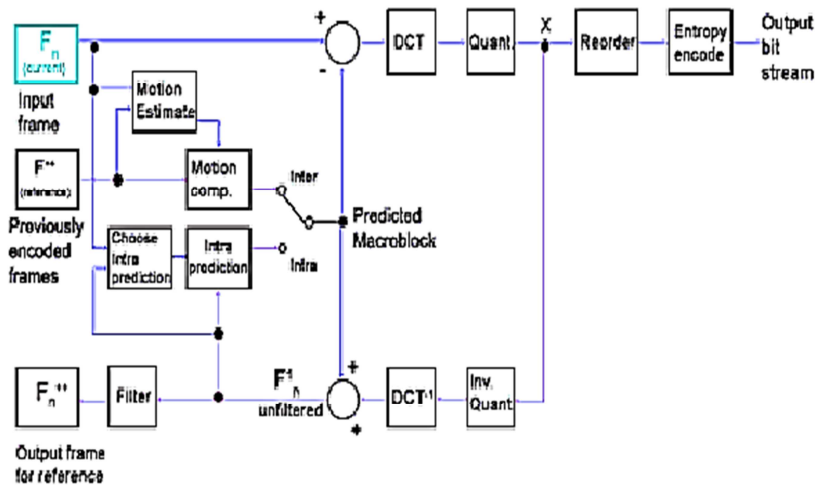


Figure 65 : MPEG 4 video coding to be used as the source coder for this JSCC scheme

There are various video codecs available which can actuate lossless video encoding on MPEG4-AVC encoding. The coding schemes differ in their efficiency, governed by the compression ratio achieved and the channel loss. There is a tradeoff between the extent of compression and the loss.

Number of information sources in a MPEG frame

For each source, the number of frequencies per source to be considered for reconstructing the voice signal (applicable for a telecom network) is governed by detection of the fundamental frequency and the semitones and the harmonics.

Each colour pixel carries a specific frequency in the phase angle domain, which is latter on converted to the frequency time domain by Inverse Fourier Transform.

For attaining a high voice quality, if we take 16x16 resolution for each voice source, then the total number of voice sources that can be catered by a single Mpeg frame (considering that the resolution supported by the MPEG frame is 1024x 1024 is 64.

Hence with a bit rate of 4 MBPS, the number of voice sources that can be accommodated (rendering a high voice quality by carrying 256 frequencies per user) is 64.

6.4. Conclusions

As the mobile network evolves to offer rich communication suite of services on top of the Voice over LTE network, the codecs used plays a vital role today as technology differentiators. In 3G mobile networks, we had narrow band AMR (Adaptive Multi Rate). In VoLTE, we have wideband AMR and EVS (Enhanced Voice Services). In this chapter, we introduce a new concept for vocoding based on JSCC using colour signals. It can have many benefits over standard vocoding methods. Though a comparison with the state of the art is not in the scope of this chapter, we opine that future research should be carried on in these lines focussing on the implementation feasibility , integrability and backwards compatibility.

8

Conclusion and Future Work

7.0. Introduction

The key issues with the available network topologies are as follows

The Network itself

- Hugely complex process of Mobility Management and Radio resource management .Basic philosophy remains the same across all the generations (2G/3G/4G).
- Complexity in radio network design. Network efficiency depends much on the frequency allocation and the frequency reuse pattern defined by the operator.

Modulation Scheme :

- Commencing from BPSK (Binary Phase Shift Keying) in 2G and ending with 64 QAM (Quadrature Amplitude modulation) in LTE -Advanced, we try to trans-receive more symbols per unit of time.
- But constrain is the Shannon bound (ref. Shannon law) of the permissible Signal to Noise ratio. The more the data rate, lesser in the inter symbol spatial distance (in complex plane), more it is prone to channel impairments and errors.
- So in LTE, the handset needs to be closer to the Base Transceiver. Augmenting coverage implies adding more base stations. Hence the concept of ENodeB and Home ENodeB has been proposed in 4G.
- So it is time to look for a smart modulation scheme to realise a smarter network.

Dependence on Layer 7 (application layer according to OSI model) processes:

- Network tracks the handset continuously.
- Handset initiates a Location Update process in a given time periodicity.
- Network and Handset manage the handover to another cell when location is changed.
- All these network activities are managed by Application layer signalling processes at the radio network.

- So the network equipments and the handset need to be intelligent enough to compute and process continuously the signalling messages for mobility and location management.
- The network equipments should be able to actuate these processor intensive activities, which increases the cost of overall network .
- For the handset as well, it is the same issue as above. Moreover, these processes drain lot of battery power, just to maintain its 'presence' in the network.
- Signalling messages at the radio layer consumes substantial bandwidth, which could otherwise have been used for data traffic.

So our primary objectives are :

- Render more intelligence to the lower layers so that it can simplify and actuate some processes (like addressing a Mobile Node), which are today layer 7 functionalities.
- Realise a smart modulation scheme which does not just aim to inject more symbols per unit time. It rather (also) attempts to identify the users at layer 1 (physical layer).
- Reduce the Mobility Management and Radio Resource Management functionalities, with a unique Access Network topology which is compatible with this smart modulation scheme.

7.1. How do we aim to achieve the objectives :

7.1.1.Propose a Hierarchical Smart Modulation scheme

- With the evolution of Mobile technology, we attempt to inject more symbols (data) per unit time.
- User identification is actuated by higher layers (of OSI)
- Our Smart Hierarchical modulation scheme has 2 concentric rings. Outer ring is meant for identifying a user. Inner ring is meant for carrying the traffic for the user. Hence we render intelligence to the physical layer.

7.1.2.Propose a smart Cell Design

- With the present topologies, the available frequency band is subdivided in Multiple sub frequency bands. Each frequency is carefully allocated to different cells in the network, taking care that no same frequency is allocated in neighbouring cell. Frequency allocation within all the cells of the network has to be meticulously done. It is hugely a resource intensive activity. In GSM, for example, the available bandwidth is divided in 125 sub frequency bands, which needs to be allocated carefully to the cells.[12].
- Handover of a call when the subscriber is moving from one cell to another is a process that requires lot of signaling interaction between the handset and the network.
- With the smart cell design we have :
 - Simple checker board design.

- The available bandwidth is divided in just 2 sub bands (against 125 of them as in GSM).
- Square shaped cells
- Handover process is driven more by the handset and is simplified.

7.1.3. Propose a Smart Network Design

- For a Closed User Group Mobile Network (say like corporate networks, petroleum refineries, factories), we do not need a network ! The intelligent handsets, along with unintelligent repeaters (acting as Access points) will form the network. For security and lawful interception capability, we however need a network component.
- For a Global Mobile network, we need an intelligent central processor which will act as a bridge between the radio and the core network.
- At the core network leg, the intelligent central processor will execute the mobility management, roaming , authentication , authorisation and other associated network processes by interacting with other network adjuncts.
- At the radio network leg, intelligent central processor will be associated with layer 1 &2 level activities and directly take part in analysing the frame at the ingress and help in the frame formation at the egress.
- Compared to the complexity we see it is analogous network components of the available technologies (MSC in 2G/3G, CSCF in IMS , MME in LTE) , we can infer that the Intelligent Central Processor is rather a light weight application , as it has to do much less application layer processing.

The key drivers for the proposed technology are:

- Reduces the cost and complexity of network setup and operations.
- Makes the handset internal design simpler. Lesser electronics and firmware will make it more affordable.
- Makes mobile communication more affordable, which is in turn, helps more penetration.
- Time to set up a network reduces , as cell / network planning part gets simplified.
- As network processes are simplified , more processing power will be available at the handset to fuel next generation intelligent applications.

7.2. Summary of the present issues with the existing topologies :

- a) The network processes, especially the mobility and location management processes in of the State-of-the-Art are intricate.
- b) The network and the handset continuously interact to actuate the intricate layer 7 processes like Location Management, Mobility Management, Radio resource management. It consumes processing power, bandwidth, needs intelligence in the network and handset (Mobile Station), drains Battery power in Mobile Station.

- c) we should thrive to realise a communication system which has least dependence on the intelligence of radio network itself , but can achieve the same level of mobility and optimisation of radio resource.

Inference

- a) The complexity existing in the various generation of the mobile networks is due to the fact that we do not witness significant improvisations on the top of the basic OSI model.
- b) So according to the structure of OSI model followed by all kinds of wireless / mobile networks, the physical layer (layer) carries information for all users. Routing and identifying a user in the network are controlled by the higher layers. Hence you require the intelligent network devices in between which can actuate these operations.
- c) So while the telecom researchers are striving to achieve more data speed (by trying to find out ways to inject more symbols, ie data per unit time (4G for example) , I was motivated to deviate from this philosophy and to aim to actuate a radical drift from the existing top heavy network architecture.
- d) In the State-of-the-Art Network, the primary drivers for the complex processes like Mobility and Location management , are meant for locating , identifying and addressing a mobile user in a network.
- e) If we have a way to actuate user identification (in the network) by using lower layer processes, we can simplify or eliminate the higher layer complex processes and hence reduce the intricacy and cost of the mobile network elements and the handsets.
- f) Every 30 minutes, the handset initiates a control message to update its location towards the Network. Handover process also requires substantial control channel message over the so called 'signalling channels'. All these actions are for maintaining its presence and to actuate the basic operations, like controlling mobility, making / breaking a call etc. This consumes processing power, signalling resources and also drains out battery.
- g) Hence we proposed a modulation scheme, part of which is directly related to the addressing function. Presently , for all mobile communication systems modulation schemes are designed only for carrying user data .

7.3. Parallelism

- a) So the big question is: Is it possible to realise a communication system without an intelligent radio network? Is it possible to have a mobile network with passive signal repeaters or alike devices?
- b) Let us take a simple example : Radio call systems for the cabs
- c) There is no network. Just a central transceiver at the Cab station.
- d) It is a broadcast network. The cab chauffeur waits to pick up the name, number, location etc for his turn. This information is processed by the chauffeur's brain, and not by the network. If you needed the information to be processed by the network, such that the information relevant to the chauffeur is only channelised to him, then we would have needed some intelligence somewhere in the network to do this. Let us call this intelligent entity sitting in the network as MR X, and let us call the chauffeur MR Y.
- e) The fundamental difference between the present cellular topologies and the proposed one is as follows.

- f) All present mobile network (comprising of the active network components in the radio and core layer) act as MR X. So he has to sit somewhere in the network. His real analogue in a mobile network: the radio and core network adjuncts (BTS/BSC/Mobile Switch etc). Specifically in 4G, they are like the eNodeB , MME (Mobility Management Entity) , SGW (Serving Gateway Entity) etc.
- g) With our proposed methodology, the active network component acts as MR Y sitting behind his radio trunking set in the car. From network perspective, it relates to the Handset and the transmission medium. No need of network intelligence active network components.
- h) Off course you still need the core network processes for Authentication, Authorisation, Accounting (Charging, billing) and all the supplementary services (CLI, call conference, SMS, MMS) and VAS.
- i) So, it is time to think behind the conventional topologies.
- j) For a network with limited mobility and closed numbering scheme, it is possible to build up a network quickly with passive devices.
- k) For the full scale mobile network, the architecture and the network processes are simplified to a considerable extent.

7.4. Prognosis

- a) Simplifying the network is one of the key drivers for 5G. Even a popular source like the Wiki says so, and so believes the telecom community.
- b) An excerpt here from Wiki (prognosis section) is as follows :

If 5G will come true and reflect these prognoses, the major difference from user point of view between 4G and 5G techniques must be something else than increased [maximum throughput](#), for example lower battery consumption, lower outage probability (better coverage), high bit rates in larger portions of the coverage area, cheaper or no traffic fees due to low infrastructure deployment costs, or higher aggregate capacity for many simultaneous users (i.e. higher [system level spectral efficiency](#)).

- c) With the advent of intelligent handsets, we tend to become more agnostic to the services offered by the network, but more dependent on the services offered directly by the OEMs like Apple, Nokia.
- d) When we use Skype (or other Over the Top services), we see Skype as a service and do not think about the internet provider.
- e) The next generation smart network technologies will permit that. With the present day technologies, this is not possible, as there is still heavy dependence on the mobile network just for rendering the basic services like call or SMS.
- f) If we are able to relieve the network from these activities by some smart topology, and make the lower layer more intelligent, then we can utilize this delta processing power in the handset to fuel the next intelligent applications.

7.5. Advantages

Less control signaling in the radio layer implies:

- a) Catering the growth and penetration of Mobile Subscribers by realization of more Micro /Pico cells with substantially lesser CAPEX & OPEX and network planning. Ideal for dense demography.

- b) Less wireless backhaul transport cost and media transport cost.
- c) Less intricate radio network design means : faster network roll out, less resources / manpower required to setup or run the network
- d) Smaller lighter handset, less expensive, as it requires lesser electronics.
- e) Per user cost decreases.
- f) Lesser battery and processing power required in the handset for basic network operations.
- g) The implementation can be done also in a very localized manner with minimum network setup cost. Like for hospitals, corporate etc.

7.6. Conclusion

It is imperative that with the changing landscape of the mobile telecommunication industry, the networks need to cope with the ever increasing volume of mobile users. Some of them are human users, some of them are integrated with machines communicating with their peers at the other end. We also have mobile networks for the Intelligent Vehicle Communications. The State-of-the-Art mobile networks are poised to meet the bandwidth demand. But more emphasis should be imparted to simplify the network architecture and the processes. With an all IP network as we see in case of 4G, the network architecture had been simplified reasonably. But there is no major shift or simplification of the network processes, especially mobility management. At present, mobility management is explicitly in the domain of the application layer. We propose to shift from this paradigm and knit this functionality with the physical layer. This entails simplification of network architecture and associated processes and helps in attaining a network which is location and presence agnostic. The interaction between the device and the network occurs only when one of them initiate a network operation for a call, SMS or other services. Exchange of Location updates between the network and the mobile device is eliminated. The handover operation is also simplified. The immediate benefits are: boost bandwidth optimisation, power utilisation and performance of device/ network processors. In course of time, the tele-density will increase even more resulting in smaller cells. Considering the state-of-the-art topologies, this will in turn induce more signalling messages due to the mobility factor. However with the smart network topology, we aim to trim off the mobility management processes and the related messages in an endeavour to streamline the performance of the overall network

With the typical M2M and H2H implementations that are foreseen, we need to attain a breakthrough in (battery and RF) power management in the mobile devices. Thanks to nano technology, there is significant progress in this domain. But this needs to be further endorsed by the enhancements in mobile network itself, by moderating the network interactions between the handset and the network.

To summarise:

- If we have a method to actuate user identification (in a mobile network) for the purpose of mobility management by implementing lower layer processes, we can simplify or eliminate the layer 7 processes and hence reduce the intricacy and cost of the mobile network elements and the handsets. This has been realized through SMNAT
- This implies that the mobile devices will derive more processing and battery power required for catering the next generation applications.
- The cost of network elements of the core and access networks will reduce as the dependence on application layer processes are diminished.

- A mobile network based on smart location management and addressing scheme provides an elegant solution to address these basic needs and helps in fostering this new breed of nano mobile devices.
- The access network design is simplified with less resource hungry processes associated to mobility management. This aligned with our endeavour to reduce the carbon footprint of the mobile networks.

9

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Appendix 1

Probability of symbol error,

$$P_s \approx 2Q(\sqrt{2A/N_0} \times \sin(\pi/M)) = 2Q(2\gamma_s \sin(\pi/M)) \quad [28]$$

Where the Q function, $Q(z)$, is defined as the probability that a Gaussian random variable x with mean 0 and variance 1 is bigger than z :

$$Q(z) = p(x > z) = \int_z^{\infty} \frac{1}{\sqrt{2\pi}} e^{-x^2/2} dx. \quad [29]$$



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